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TECHNICAL INFORMATION BULLETIN 93-13

TRANSPORTING VIDEO TELECONFERENCING TRAFFIC

DECEMBER 1993

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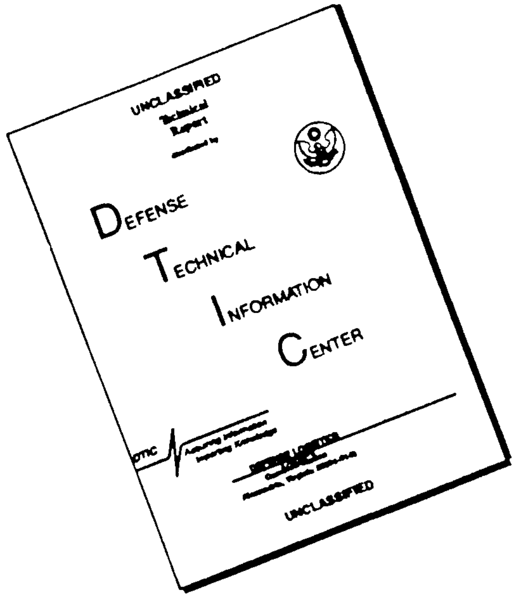

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The purpose of this document is to summarize the work accomplished on Subtask 4 (Transporting Video Teleconferencing Traffic) of Task 2 (Technical Work in the Area of Videoconferencing) of this contract. The purpose of this subtask was to study alternative media for transporting video teleconferencing traffic. In the first step of the project basic transport media (as opposed to Alternative Media) which are commonly used for video conferencing were reviewed. This analysis provides reference for the discussion of the non-traditional telecommunication techniques. Four alternative transport media were examined: LANs/MANs, PSTN, mobile radio, and Internet. Each of these media is reviewed and analyzed in a separate section of the report. The status of the technology for each medium is reviewed including a review of the standardization status. Commercial products, if any, are identified and classified. Finally the applications and implications of the medium to provide video telephony services in the Government community are reviewed. Overall conclusions and recommendations are included.

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TASK NO. 2
TECHNICAL WORK IN THE AREA OF VIDEO TELECONFERENCING

SUBTASK 4
TRANSPORTING VIDEO TELECONFERENCING TRAFFIC

FINAL REPORT
CONTRACT DCA100-91-C-0031

Submitted to:
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WASHINGTON, DC

November 1, 1993

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NCS TECHNICAL INFORMATION BULLETIN 93-13

TRANSPORTING VIDEO TELECONFERENCING TRAFFIC

DECEMBER 1993

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FOREWORD

Among the responsibilities assigned to the Office of the Manager, National Communications System, is the management of the Federal Telecommunication Standards Program. Under this program, the NCS, with the assistance of the Federal Telecommunication Standards Committee identifies, develops, and coordinates proposed Federal Standards which either contribute to the interoperability of functionally similar Federal telecommunication systems or to the achievement of a compatible and efficient interface between computer and telecommunication systems. In developing and coordinating these standards, a considerable amount of effort is expended in initiating and pursuing joint standards development efforts with appropriate technical committees of the International Organization for Standardization, and the International Telegraph and Telephone Consultative Committee of the International Telecommunication Union. This Technical Information Bulletin presents an overview of an effort which is contributing to the development of compatible Federal, national, and international standards in the area of Video Teleconferencing. It has been prepared to inform interested Federal activities of the progress of these efforts. Any comments, inputs or statements of requirements which could assist in the advancement of this work are welcome and should be addressed to:

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National Communications System
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APPENDICES

1.0 INTRODUCTION

The National Communications System has awarded Contract DCA 100-91-C-0031 entitled "System Engineering and Technical Assistance in the Area of Facsimile and Video Teleconferencing for the Office of the Manager", to Delta Information Systems. The purpose of this document is to summarize the work accomplished on Subtask 4 (Transporting Video Teleconferencing Traffic) of Task 2 (Technical Work in the Area of Videoconferencing) of this contract.

The purpose of this subtask was to study alternative media for transporting video teleconferencing traffic. In the first step of the project basic transport media (as opposed to Alternative Media) which are commonly used for video conferencing were reviewed. This analysis, contained in section 2.0 provides reference for the discussion of the non-traditional telecommunication techniques.

Four alternative transport media were examined: LANs/MANs, PSTN, mobile radio, and Internet. Each of these media is reviewed and analyzed in a separate section of the report. The status of the technology for each medium is reviewed including a review of the standardization status. Commercial products, if any, are identified and classified. Finally the applications and implications of the medium to provide video telephony services in the Government community are reviewed.

Overall conclusions and recommendations are included in Section 7.0.

2.0 BASIC TRANSPORT MEDIA FOR VIDEO TELECONFERENCING

In general, video teleconferencing requires a high transmission bit rate relative to other services such as voice and data. For that reason, the availability of teleconferencing for the government community is dependent upon the availability of ubiquitous, inexpensive communication channels operating at high bit rates. The purpose of this section is to review the primary telecommunication options which are now used for Video TeleConferencing (VTC) today. The discussion is divided into four parts: (1) Non - ISDN Media (2) narrow band ISDN, (3) broadband ISDN, and (4) FTS-2000.

2.1 Non - ISDN Transport Media for VTC

Today's teleconferencing user has a range of choices in transmission, both in network options and selection of data rate. Customarily, teleconferencing transmission rates are divided into two general categories -- narrowband (switched 56 kbps circuits) and wideband (384 kbps, 768 kbps, 1.544 Mbps) -- though even this distinction has blurred somewhat as new technologies enable the delivery of wideband transmission rates through aggregations of narrowband circuits.

Typically, wideband VTC networks are implemented either through dedicated private T1 networks or virtual private networks (both available from all major carriers), or through the use of carriers' reservation-based switched wideband networks (AT&T's Accunet Reserved, Sprint's Meeting Channel, and MCI's VideoNet). Transmission rates are normally 384 kbps, 768 kbps, or 1.5 Mbps. It is not uncommon for a user to maintain a private network for internal videoconferencing while periodically using the carriers' switched services for communication with off-network sites -- particularly international connections.

In either a dedicated or switched environment, a dedicated T1 trunk must be brought to the user's premises. If the user wants to retain the flexibility of choosing services from multiple carriers, then the access line will be routed through the local telephone company central office; otherwise, the access line is connected to a single carrier's point of presence for use either in a dedicated or switched configuration.

Narrowband VTC transmission relies on switched-56 digital service offerings. Switched-56 network services became accessible to VTC users in 1987, when two 56 kbps circuits could first be synchronized for a total data rate of 112 kbps. A single switched-56 circuit is generally inadequate for video teleconferencing, but two 56 kbps circuits can provide enough bandwidth to accommodate the 16 kbps

or 32 kbps that must be assigned to audio while leaving sufficient capacity for video transmission.

While the video quality is obviously reduced relative to that available in wideband videoconferencing, many companies have found it to be acceptable for a range of teleconferencing applications. These companies have found that the trade-off of lost picture quality is compensated for by the inexpensive transmission costs for 56 kbps dial-up circuits. In 1985, the cost of a switched 56 kbps line war around $75 per hour; today, a switched domestic conference at 112 kbps will cost no more than $26 per hour -- not much more than a normal long distance phone call.

Additionally, the introduction of inverse multiplexing technology into the videoconferencing market has radically altered users' options for increased bandwidth. Inverse multiplexers (I-muxes) permit wideband videoconferences to be carried on switched-56 networks; they do so by dividing a wideband data stream emanating from a codec into multiple individual narrowband streams -- literally, by performing the inverse function of a standard multiplexer. At the receiving end, an I-mux accepts the multiple narrowband data streams and synchronizes them into a single wideband signal for receipt and processing by the video codec.

An inverse multiplexer enables users to videoconference over dial-up digital circuits at rates up to 1.344 Mbps (24 x 56 kbps). This allows users to retain the flexibility and low cost of dial-up access while enabling better picture quality through higher data rates.

If users only employ switched-112 services, then the access lines are 56 kbps circuits. Use of an inverse multiplexer requires users to lease T1 access to the carrier's point of presence. (The cost of T1 access is usually equivalent to four 56 kbps circuits, however, so it is often easy to justify.) After access charges, the user pays only for the actual calls placed on the switched network.

The Department of Defense has established the Defense Communications Teleconference Network (DCTN) to provide teleconference services within that agency. This network provides switched service at a transmission bit rate of 1.544 mbps. Work is presently underway for reducing this bit rate to 768 or 384 Kbps.

2.2 Narrowband ISDN

Users looking for improved network performance (i.e. faster call set-up, better connections between sites) or access to high-speed switched data services may use ISDN for videoconferencing. ISDN (Integrated Digital Services Network) refers to an emerging set of worldwide standards for end-to-end digital information transfer. It is endorsed by the International Telecommunications Union (ITU), the American National Standards Institute (ANSI), and virtually every other national telecommunications authority. Eventually, ISDN will transform the present telephone network into an end-to-end switched digital network providing variable transmission rates in 64-kbps increments.

A primary objective of ISDN is to provide a variety of network services through a standard set of network interfaces. This will enable users in most cases to access voice, data, and image via a single standard network connection. A second objective, interoperability of switches and terminal equipment, will be assured by conformance to standards. Such standardization will lead to significant reductions in equipment costs; ISDN will also contribute to reduced operating costs through its capability of continuously monitoring loop performance over the embedded maintenance channel.

While Japan and some European countries have deployed ISDN more extensively, fewer than 40% of U.S. central offices are currently equipped to handle ISDN traffic. Growth should be steady, however; the Regional Bell Operating Companies (RBOCs) project that by the end of 1994, they will have almost 65 million ISDN-capable lines (57% of all lines) and 1,832 ISDN-capable switches (19% of all switches). Additionally, each of the seven RBOCs has put ISDN basic rate tariffs in place.

The Federal Government also strongly endorsed ISDN by awarding a ten-year, multibillion dollar contract for FTS-2000 to A&T and Sprint. Both proposals were based on ISDN architecture.

2.2.1 Basic Rate Interface (BRI)

The Basic Rate Interface (BRI) is comprised of two 64-kbps data channels, know as B channels, and a single 16-kbps signaling channel known as the D channel. The data and signaling channels are time-division multiplexed into a single stream to provide a composite full-duplex bandwidth of 144 kbps.

BRI provides a composite bandwidth of 144 kbps (2 x 64 kbps B channels + 1 x 16 kbps D channel) and full duplex transmission with time division multiplexing

(TDM) into a single stream containing both user and signalling information.

Basic rate ISDN uses existing network facilities, typified by the twisted pair local loop and 64-kbps digital switch. Basic rate service (2B+D) can be provided to the customer by replacing and analog telephone with an ISDN telephone and a network termination. The B channels (64 kbps) provide circuit-switched voice and data and packet-switched data; the D channel (16 kbps) provides out-of-band signalling plus packet data. Basic rate ISDN allows applications such as Group IV facsimile, PC screen sharing, image transfer and limited motion videophone. (Primary rate (2B+D) provides up to 1.536 Mbps.)

All seven Regional Bell Operating companies have filed ISDN Basic Rate Interface (BRI) tariffs.

As inter-switch SS7 connectivity becomes pervasive and equipment for applications becomes less expensive, ISDN will become more acceptable to the small user.

2.2.2 Primary Rate Interface (PRI)

The Primary Rate Interface (PRI) is comprised of 23 64-kbps B channels, and one 64-kbps D channel for signaling, for a composite bandwidth of 1.536 Mbps. (In Europe, the ISDN PRI configuration is 30B + D.)

Under the PRI standard, certain aggregations of B channels can create high-speed H channels. Six PRI B channels define an H0 channel at 384 kbps; 23 B channels define an H10 channel at 1.472 Mbps; 24 B channels define an H11 channel at 1.536 Mpbs. These wideband H channels could possible be used to meet most demands of data communications users.

Primary rate ISDN (PRI) consists of 23 64-kbps B channels (30B channels in Europe) and one 64-kbps D channel for signalling. The PRI may also be allocated as high-speed (H) channels at 384, 1472 or 1536 kbps. The H channels are currently viewed as wideband circuits and could possibly be used to meet most demands of data communications users.

Primary rate channels can be used to multiplex lower data rate channels, provide high-data-rate wide area network connectivity and implement private branch exchange (PBX) connectivity.

PRI can also be used for LAN-to-LAN connectivity.

2.3 Broadband ISDN

The Integrated Services Digital Network (ISDN) is divided into two parts -- narrowband and broadband. Narrowband ISDN, described in the last section, operates at rates equal to or less than the primary rates (e.g. 1.544 Mbps). Broadband ISDN, also known as B-ISDN, operates at transmission speeds above the primary rate. B-ISDN promises to allow very high speed digital transmissions (155.52 Mbps, 622.08 Mbps, 2488.32 Mbps) that will dwarf the N-ISDN.

While the standards for N-ISDN have been sufficiently set in place to permit deployment of N-ISDN networks and applications today, B-ISDN is not yet market-ready. Because B-ISDN is still nascent, this section necessarily concentrates on providing an update of relevant standards and definitions of B-ISDN rather than describing the technology in terms specific to videoconferencing.

One backbone structure for the B-ISDN consists of a new Synchronous Digital Hierarchy (SDH) defined in CCITT Recommendations G.707, G.708, and G.709[1][2][3]. These recommendations are concerned with bit rates for the synchronous digital hierarchy, details of the resulting network node interface, and a synchronous network multiplexing structure. Three digital Synchronous Transfer Modes (STM) have been so far specified:

- STM-1 : 155.52 Mb/s
- STM-4 : 622.08 Mb/s
- STM-16: 2488.32 Mb/s

The North American version of the standards uses a basic transport module of 51.84 Mb/s (Synchronous Transport Signal-Level 1, STS-1) and as described by Ballart and Ching[4].

In addition to optical interfaces for the above synchronous bit rates (SONET - Synchronous Optical Network), CCITT Working Party XVIII/7 has agreed to standardize an STM-1 electrical interface for inclusion in Recommendation G.703. SONET is defined in the U.S. by the ANSI standards as listed below.

T1.105-1988	Optical Interface Rates & Formats
T1.106-1988	Optical Interface Specifications: Single-mode
T1.105-1990a	Addendum (in voting process)
T1.xxx-1990	Optical Interface Specification: Short Reach (in voting process)

Figure 2.1 compares the new synchronous digital hierarchy with the existing digital hierarchy.

The general structure and service capabilities of the ISDN are defined in a total of 77 CCITT Recommendations I.110 through I.605. Within this set, Recommendation I.121 provides the basic description of the B-ISDN. Much of the material in the following paragraphs is derived from this Recommendation.

Asynchronous transfer mode (ATM) is the target transfer mode solution for implementing a B-ISDN. It will influence the standardization of digital hierarchies and multiplexing structures, switching and interfaces for broadband signals.

ATM concerns a specific packet-oriented transfer mode using the asynchronous time division multiplexing technique: the multiplexed information flow is organized in fixed size blocks, called cells. A cell consists of a user information field and a header; the primary role of the header is to identify cells belonging to the same virtual channel on an asynchronous time division multiplex. Cells are assigned on demand, depending on the source activity and the available resources. Cell sequence integrity on a virtual channel is preserved by the ATM layer.

ATM is a connection-oriented technique. Header values are assigned to each section of a connection when required and released when no longer needed. The connections identified by the headers remain unchanged during the lifetime of a call. Signalling and user information are carried on separate virtual channels. ATM will offer a flexible transfer capability common to all services, including connectionless services.

Conversational Services

Conversational services in general provide the means for bidirectional dialogue communication with real-time (no store-and-forward) end-to-end information transfer from user to user or between user and host (e.g. for data processing). The flow of the user information may be bidirectional symmetric, bidirectional asymmetric and in some specific cases (e.g. such as video surveillance), the flow of information may be unidirectional. The information is generated by the sending user or users, and is dedicated to one or more individual communication partners at the receiving site. Examples of broadband conversational services are videotelephony, video conference and high speed data transmission.

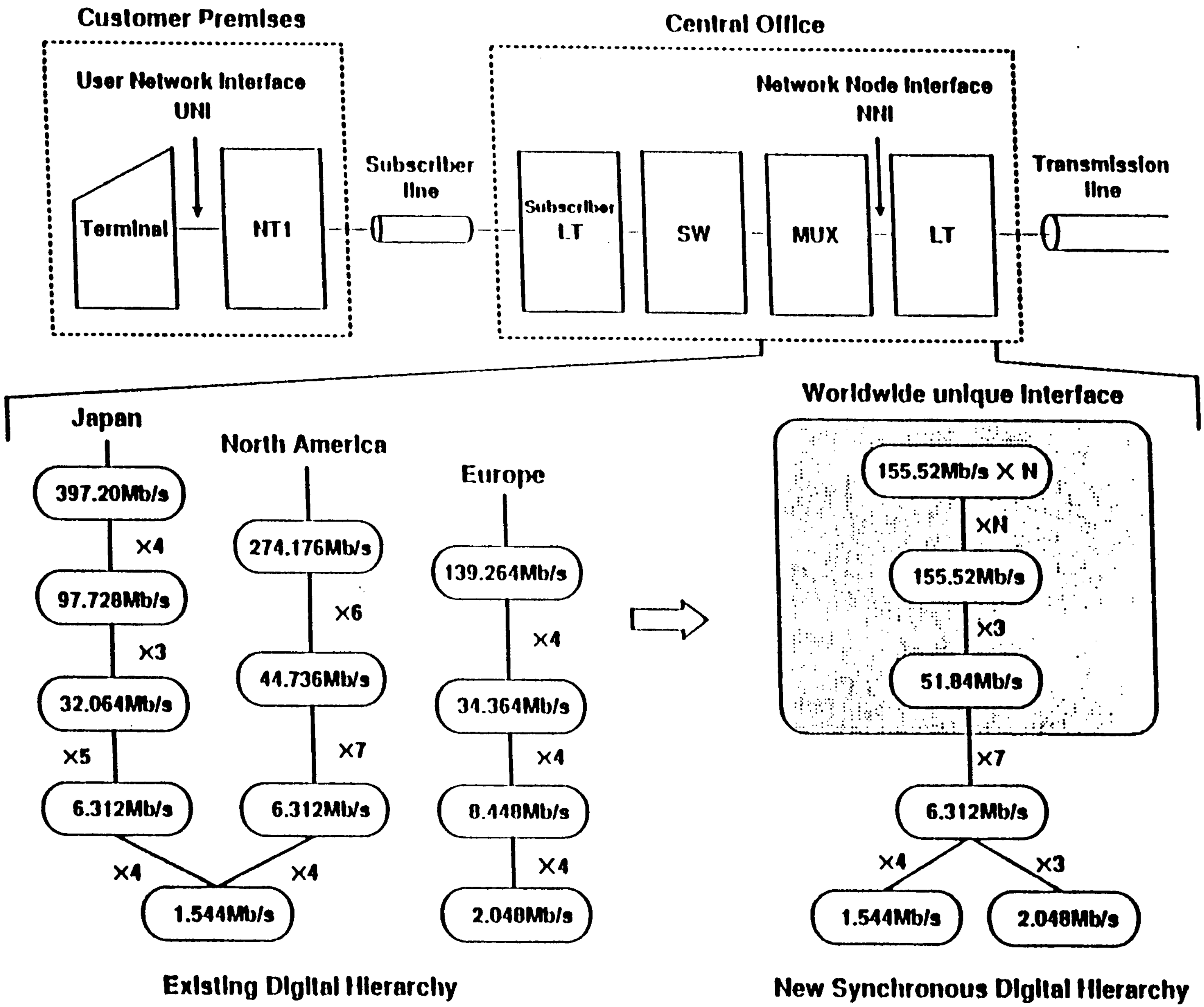

FIGURE 2.1
COMPARISON OF NEW SYNCHRONOUS DIGITAL HIERARCHY TO EXISTING DIGITAL HIERARCHY

Information Flows

Video service information can be characterized in many ways, including:

- The direction of information flow: video services may be bidirectional, e.g. videotelephony and videoconference, or essentially unidirectional, e.g. video distribution services for business and entertainment.
- The symmetry of information flow: messaging, retrieval and distribution services are characterized by asymmetrical information flows.
- The origin of the source material: how video signals enter the network (e.g. direct from camera, from storage media, via satellite or other delivery mechanisms) can also provide a means of characterizing service information flows.

The B-ISDN will be based on ATM techniques which are well suited to supporting source traffic which is time varying. The establishment of virtual connections which involve the transfer of information only when required will mean that the resources of the network can be closely matched to the needs of the source traffic.

ATM Performance Parameters which are of concern for the VTC application core.

- Cell Loss Ratio
- Cell Error Ratio
- Cell Transfer Delay
- Mean Cell Transfer Delay
- Cell Delay Variation

2.3 FTS-2000

The FTS-2000 Project was initiated by the General Services Administration in the mid-1980's to provide standard common carrier telephone network services to Government agencies that would satisfy other than specialized information resources needs. Contracts were awarded in 1988 to two common carriers, AT&T and SPRINT for FTS-2000 services. The networks operated by these carriers are referred to as Network A and Network B, respectively. Individual agencies are assigned to either network in accordance with a plan to maintain a specific

percentage traffic distribution between the networks at the time of the 1988 award.

The services provided to users have been expanded. Currently Switched Data Services (SDS) and Dedicated Transmission Services (DTS) are available. Compressed Video Transmission Services (CVTS) is now being provided on the networks. Its inclusion requires specific requests for the service from user agencies.

REFERENCES

[1] CCITT Recommendation G.707, "Synchronous Digital Hierarchy Bit Rates," Study Group XVIII Plenary Assembly, Melbourne, Nov. 1988.

[2] CCITT Recommendation G.708, "Network Node Interface for the Synchronous Digital Hierarchy," Study Group XVIII Plenary Assembly, Melbourne, Nov. 1988.

[3] CCITT Recommendation G.709, "Synchronous Multiplexing Structure," Study group XVIII Plenary Assembly, Melbourne, Nov. 1988.

[4] R. Ballart and Y.C. Ching, "SONET: Now Its the Standard Optical Network," *IEEE Commun. Mag.*, pp. 8-15, Mar. 1989.

3.0 VIDEO TRANSPORT VIA LANS/MANS

3.1 LAN/MAN OVERVIEW

A local area network (LAN) is a communications network that provides interconnection of a variety of data communicating devices within a small area. The most common occurrence is a network that is confined to a single building. Networks that span several buildings, such as on a college campus or military base, are also common. Another element that could be added to the definition is that a local network is generally privately owned rather than a public or commercially available utility. Indeed, typically, a single organization will own both the network and the attached devices. Some of the typical characteristics of local networks are:

- single organization proprietorship
- distances involved are of the order of a few miles and in the general locality
- the deployment of some type of switching technology
- a transmission medium is shared among the attached devices
- transmission is in the form of packets
- a transmission from any one station is received by all other stations (hence the term "packet broadcasting"
- there is no master station; rather, all of the stations cooperate to assure orderly use of the transmission medium
- high data rate (0.1 - 100 mbps)
- low error rates (10^{-8} to 10^{-11})

In recent years, a new type of network, referred to as a **metropolitan area network (MAN)**, has been developed. A metropolitan area network shares the characteristics listed above with the LAN; the difference is that the MAN covers larger distances and, generally, operates at higher data rates.

The LAN and MAN are distinguished from **wide-area networks (WANs)**. As the name implies, WANs are networks that cover substantial distances. Public telephone networks and packet-switching networks are examples of WANs. Table 3-1 summarizes some of the key characteristics of LANs, MANs, and WANs.

The IEEE Standards organization has assumed responsibility for the development of LAN and MAN standards. The standards which have been developed are illustrated in Figure 3.1.

<table>
<tr><td colspan="7">802.2 Logical link control</td><td>LLC</td><td rowspan="2">2</td></tr>
<tr><td rowspan="2">802.3
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token bus</td><td rowspan="2">802.5
token ring</td><td rowspan="2">802.6
**DQDB</td><td rowspan="2">802.7
broadband</td><td rowspan="2">802.8
fiber</td><td rowspan="2">802.9
integrated</td><td>MAC</td></tr>
<tr><td>Physical</td><td>1</td></tr>
</table>

* carrier sense multiple access with collision detection

** distributed queue dual bus

OSI layer ↑

Figure 3.1
The IEEE 802 LAN Standards

TABLE 3-1 Characteristics of LANS, MANS, and WANS

Network	**Data Rate**	**Distance Covered**
Local Area Network (IEEE 802)	1-20 Mbps	< 25 km
Fiber Distributed Data Interface	100 Mbps	< 200 km
Metropolitan Area Network (IEEE 802.6)	30 Mbps-1^{+}Gps	< 160 km
Traditional Wide-Area Network	10 kbps-1.5 Mbps	unlimited
High-Speed Wide-Area Network	50 Mbps-1^{+}Gbps	unlimited

MAC and LLC

- The OSI data link layer of LANs (layer 2) is decomposed in to the media-access control (MAC) and the logical-link-control (LLC sublayers.

- The MAC sublayer regulates the access to the channel shared by nodes on a LAN.

- The LLC sublayer supervises the packet links between nodes.

- The *efficiency* of a MAC protocol is the maximum fraction of time that the nodes can transmit packets successfully when they use the protocol and when the network is heavily loaded by many nodes.

- The *throughput* of a LAN is the maximum rate of successful bit transmissions on the LAN when it is heavily loaded by many nodes. The throughput is the product of the transmission rate and the efficiency of the MAC protocol.

Networks based on the IEEE 802.3 standards compose the most widely used LANs. These are bus networks (see Figure 3.2) that use media-access control protocol called *carrier sense multiple access with collision detection* (CSMA-CD).

Token ring networks are another widely used family of LANs. The token ring networks (see Figure 3.2) were developed by IBM in the early 1980s. The transmission medium is typically a twisted pair or a coaxial cable, although some versions use optical fibers. The MAC protocol of the token ring is as follows: A specific bit pattern, called the *token*, circulates on the ring. When a node wants to transmit, it waits until the token comes by. It then replaces the token with another pattern (SFD) which indicates the *start of frame*, and it appends its packet. The node converts the token in to an SFD by monitoring the signal it receives from the ring and by modifying the token while it is stored in the interface buffer before retransmitting it. Once the packet has been transmitted, the node transmits the token, which then becomes available to another node.

It is useful to visualize the location of the IEEE 802 protocol standards in the context of other related protocol stacks. Table 3-2 compares the OSI layers to CCITT, TCP/IP, and IEEE 802. Note that TCP/IP is frequently combined with Ethernet to form a total protocol stack.

TABLE 3-2 Common Protocol Layers

	OSI	CCITT	DOD	IEEE 802
7.	Application			
6.	Presentation		Various	
5.	Session			
4.	Transport		TCP	
3.	Network	X.25	IP	
2.	Link	LAP-B		Logical link control
				Medium access control
1.	Physical	X.21		Physical

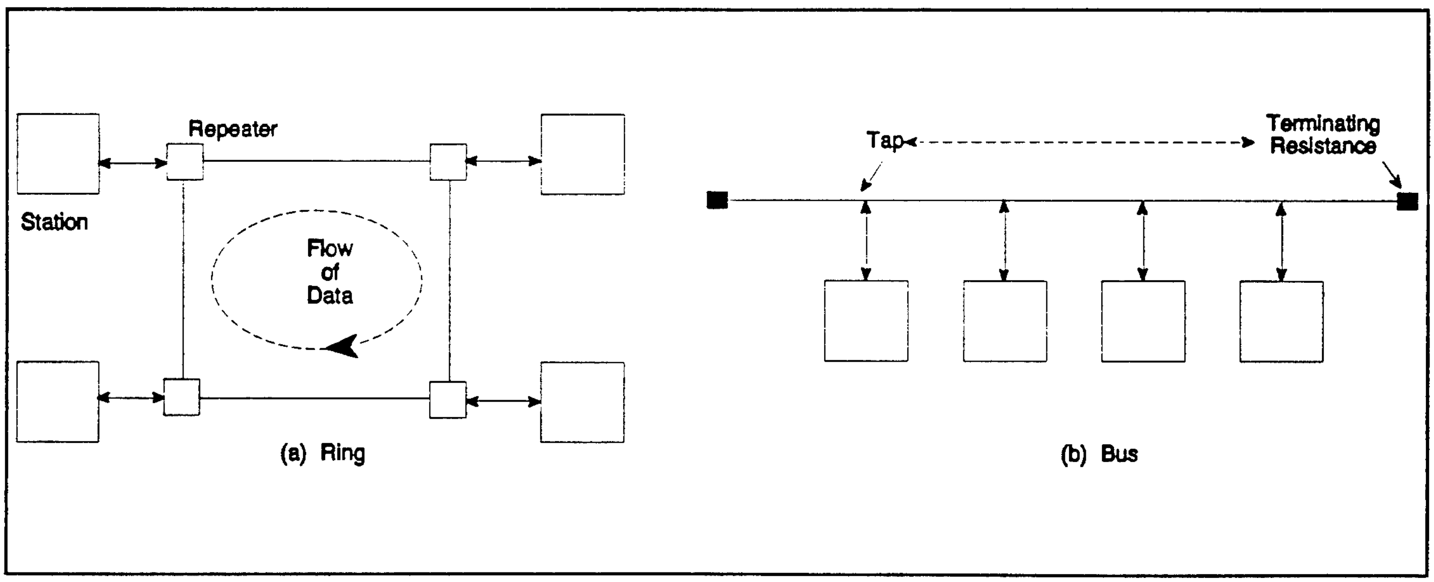

FIGURE 3.2
LAN TOPOLOGIES

Fiber Distributed Data Interface

The fiber distributed data interface (FDDI) is a network standard developed by the American National Standards Institute (ANSI) that specifies a 100-Mbps optical fiber ring network. The *fiber distributed data interface* (FDDI) network is a dual-ring network connecting nodes with a maximum length of the fiber of 200 km. In the literature, FDDI is generally considered to be a LAN and, indeed, most of the existing installations are within a single building or small cluster of buildings. FDDI is designed to provide high end-to-end throughput between expensive, high-speed devices such as mainframes and mass storage devices. It is also used as a backbone network to connect a number of lower-speed LANs. Table 3-3 is a comparison of FDDI and 802.5 parameters.

As with the typical LAN, FDDI was originally defined to use packet broadcasting and to support data traffic. A recent enhancement to the standard, known as FDDI-II, provides support for voice traffic and other applications that normally use circuit switching.

TABLE 3-3 Differences Between FDDI and 802.5

FDDI	802.5
Optical fiber	Shielded twisted pair
100 Mbps	4.16 Mbps
Reliability specification	No reliability specification
4B/5B code	Differential Manchester
Distributed clocking	Centralized clocking
Timed token rotation	Priority and reservation bits
New token after transmit	New token after receive

ATM

The Asynchronous Transfer Mode (ATM) format was developed to fulfill a WAN requirement within the B-ISDN. However, the format also lends itself for use as a wideband LAN and is beginning to find significant use in this mode. Shared media ATM networks include DQDB, the Cambridge Fast Ring, the Cambridge Backbone Ring, the Orwell Ring, and the recently proposed ATM Ring. There are a number of solutions based on interconnected switches to form general topology networks.

GENERAL VTC PERFORMANCE ISSUES FOR LAN/WAN MEDIA

Today, virtually all video teleconferencing is transmitted via synchronous transmission circuits such as switched 56 Kbps and T1 lines. LANs and WANs are fundamentally different from these networks in many ways, and some of these differences can contribute to degraded VTC performance unless precautions are taken. Some of the technical issues which must be carefully considered in the design of a VTC system via LANs are listed below.

- The throughput bitrate allocated to one particular user usually fluctuates depending upon the number of active users and overall traffic through the network. If the volume is very high, the available bitrate may be so low the quality is significantly degraded.

- Packet loss from LANs can cause a more serious degradation to the audiovisual signals than distributed bit errors associated with synchronous links.

- Occasional serious interruptions in the connection are much more likely in LANs than synchronous connections. When such problems occur, the audio signal can be rendered almost useless. It is interesting to note that the audio is far more critical in VTC sessions than the video. If the audio is unusable, the conference is unusable regardless of the video. However, it is also interesting to note that "useful" video (e.g. still pictures) can be transmitted under surprisingly adverse conditions.

- In order to achieve an interactive VTC service, the communication media must have low delay. In the LAN situation, this means that the packet rate must be high enough to achieve low latency. In many LANs where the payload bits/packet is fundamentally high, this will dictate that the bitrate for the audio is governed by the network rather than the source signal itself.

3.2 VTC Products and Applications

There is a great deal of activity in the use of LANs for video teleconferencing. A large number of commercial products are available for this application, and there is work to develop standardized terminals for VTC over LAN. work in these two areas is described below.

3.2.1 Commercial Products

Examples of commercial products for providing VTC services over LANs are listed in Table 3-4 and brochures are included in Appendix 3.1.

3.2.2 VTC Standards

At the September 1993 meeting of the ITU SG15, Mr. Okubo was appointed Rapporteur to adapt the H.320 Recommendations for operation over LAN networks. The modified Recommendations have been designated as H.32Z.

TABLE 3-4 Desktop Videoconferencing Products Using LANs

Product	Company	Product	H.261 Std	WAN Connectivity	LAN/MAN Connectivity	HDW Platform	Multipoint Support	Resolution pixels	Frame Rate Frame/sec
Invision	Intervision Systems	Board, Software	x	ISDN	-Ethernet* -Token Ring -FDDI	PC		256x240	22
Personal Viewpoint	Viewpoint Systems	Board, Software	CPV	Router	-Ethernet* -Token Ring -FDDI	PC	X	256x200	Variable 30 max
VIS-A-VIS	Worldlinx Telecomm	Software	x	ISDN	-Ethernet -Token Ring -Net Bios	PC		640x480	Variable
Connect 918	NUTS Technology		x	ISDN	Ethernet	MAC	X	352x288	15
Communique	In Soft	Software	x	ISDN 56Kbps-T1	ATM Ethernet FDDI	Sun Work-station	X	640x480	15
Communicator III	Eyetel		X	ISDN	Ethernet			352x288	15
Picture Window	BBN	Software		Internet		Sun Work-station	x	320x250	3-6

* TCP/IP Used for Higher Layers

4.0 VIDEO TRANSPORT VIA THE PSTN

4.1 Overview

The Public Switched Telephone Network (PSTN) is so universal and ubiquitous that no description is required. It provides telephone service reliably and inexpensively to literally all parts of the world. One good measure of the leverage and power of the PSTN is the recent Group 3 facsimile revolution. No one remotely visualized the potential for G3 fax because no one fully appreciated the value of a PSTN connection.

A number of recent technological breakthroughs have made it possible to realistically provide videophone service over the PSTN. These developments are listed below.

- advanced video and speech compression technology
- advanced PSTN modem technology (Up to 28 Kbps)
- development of packet transmission techniques
- low cost VLSI technology

Several companies have developed PSTN videophones. AT&T and Marconi are now actively marketing units to the consumer on a worldwide basis. All PSTN videophones which have been developed are discussed in Section 4.2.

The ITU has initiated an "urgent" program to develop an international standard for the PSTN videophone. The schedule calls for the development of a stable draft by November 1994. This standardization effort is described in section 4.3. In section 4.4 implications of the PSTN videophone for the Government community are presented.

4.2 Commercial Products and Applications

Table 4-1 is a summary of the technical characteristics of existing PSTN videophones. Marconi manufactures a unit which is marketed by British Telecom and MCI. The AT&T unit is designated as the Model 2500. The Marconi and AT&T products have been marketed extensively to the consumer in 1993. Comtech and Sharevision have developed products but not marketed as intensely. The Sharevision product is interesting because it includes an interactive data capability to make it particularly applicable to a PC workstation. The European Cost 211ter and Bosch products are being tested and evaluated. Brochures on some of these products are included in Appendix 4.1.

TABLE 4-1

Technical Characteristics of Existing PSTN Videophones

	MARCONI	AT&T	COMTECH	SHARE-VISION	COST 211TER (1)	BOSCH
SYSTEM						
Total Bit Rate (kbit/s)	9.6/14.4 kbit/s	16.8/19.2 kbit/s	< 32 kbit/s	9.6/14.4 kbit/s	9.6/14.4/ 19.2/24 kbit/s	16.4 kbit/s
Multiplexing Method	Frame Based	LAP-B	Proprietary	packetization (proprietary)	H.221-like	Packet
MODEM						
Bit Rate	9.6/14.4 kbit/s	16.8/19.2 kbit/s		9.6/14.4 kbit/s	V.FAST (3) V.32bis	14.4 - 24 kbit/s
Error Correction	Trellis coding	yes	MNP4/LAPM	No	No	RCPC
Standard	V.32 based	no	V.34/V.Fast	- V.32bis - V.FAST (3)	- V.32bis - V.FAST (3)	?
Modulation		FAST START				128 - QAM
VIDEO						
Bit Rate - kbit/s	4, 8.5 kbit/s	var	up to 32 kbit/s	9.6 kbit/s - 0 kbit/s	4.4, 9.2, 14.0, 18.8 kbit/s	8 kbit/s
Luminance Pixels/Line	128	128	96 to 320	96	176	176
Luminance Lines/Frame	96	112	80 to 240	80	144	144
Chrominance Resolution Ratio (2)	4:2:0	4:1:0	4:2:0	4:2:0	4:2:0	4:2:2
	2:1, 2:1	4:1, 4:1	4:1:1 (YUV)	4:2:1	4:2:0	4:2:2

(1) Hardware Demonstrator Model
(2) MPEG Notation
(3) It is recognized that V.FAST is not an approved standard at this time.

TABLE 4-1
CONTINUED

	MARCONI	AT&T	COMTECH	SHARE-VISION	COST 211 TER (1)	BOSCH
Frame Rate (fps)	var	var	up to 15 f/s	0-12 f/s	var	5, 6.25, 8.33 f/s
Coding Algorithm	proprietary - similar to H.261	DCT-DFD	APEX	proprietary (modified H.261+)	hybrid DCT/DFD half pixel	Bosch
Motion Accuracy	pixel	¼ pixel			½ pixel	
DCT	8x8	8 x 8			8x8	
AUDIO						
Bit Rate - kbit/s	5.0 kbit/s	6.8 kbit/s	2.4 to 9.6 kbit/s	4.8 kbit/s	4.8, 6.8 kbit/s	2.4 kbit/s
Bandwidth - Khz	3.1 kHz	4 kHz	3.2 kHz	8 kHz	4 kHz	?
Coding Algorithm	CELP	CELP+	STC	proprietary	proprietary CELP+	LPC
USER DATA						
Bit Rate	zero (4)	zero (4)	zero (4)	as needed: 0-9.6 kbit/s	zero (4)	zero (4)

(4) Provision for a separate user data channel is not provided.

4.3 Standardization Activity

In 1992 the ITU-TS appointed a Rapporteur to perform a study to determine the feasibility of developing a Recommendation for a PSTN videophone. The study was completed in September 1993 and recommended that work be initiated to develop a near term Recommendation (1995) to be followed by an enhanced version in 1998. The ITU-TS has re-appointed the Rapporteur to develop these Recommendations as detailed in Table 4-2 and Figure 4.1. (See Appendix 5.3.)

TABLE 4-2

ITU-T Recommendations for the Very Low Bitrate Videophone

FUNCTIONAL ELEMENT		NEAR TERM (1995)	LONG TERM (1998)
SYSTEM		H.32P	
VIDEO CODER		H.26P	H.26P/L
SPEECH CODER		AV.25Y	WP 15/2 (4Kbps)
DATA INTERFACE		Based On MLP or HDLL H.DLP	
SUPERVISION CONTROL		H.24P	
MULTIPLEX/ ERROR CONTROL		H.22P	
MODEM	PSTN	V.32 bis, V.34/V.8 (V.FAST)	
	MOBILE RADIO	--	FPLMTS

Recommendation will not be developed by WP 15/1.

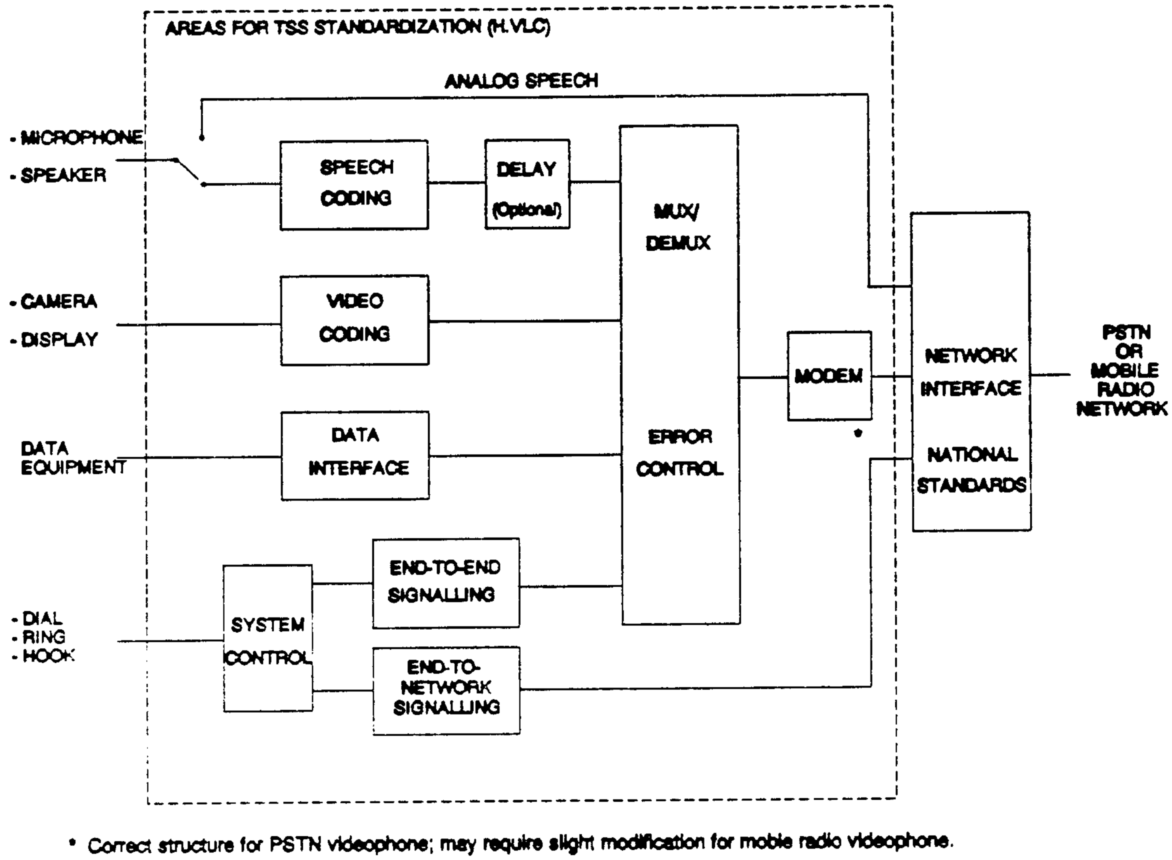

FIGURE 4.1
FUNCTIONAL BLOCK DIAGRAM FOR VERY LOW BITRATE VIDEOPHONE

The status of each of the Videophone Recommendations being developed is summarized below.

PSTN Modem

V.32bis (9.6 and 14.4 Kbit/s) and V.FAST (up to 28.8 Kbit/s) modems are mature and would be employed, thereby greatly increasing the system's performance relative to existing videophone products.

H.26P - Video Coder

The near term video codec will provide significant improvement in picture quality relative to H.261, when adapted to videophone, as demonstrated by computer simulation. Potential improvements result from sub-pel motion

compensation, reduced overhead, and motion compensation at the block level. Picture formats being considered include QCIF and lower resolutions.

H.26/L - Long Term Video Coder

The underlying source model of the current standards is: 2D rigid translationally moving blocks containing highly correlated pels. In general, this is only a very rough approximation of reality. Differences in subsequent images are caused by motion of objects, but also by camera motion (zoom, pan), camera noise, lighting effects, changes in the shape of objects, occlusion of objects and background, scene cuts, etc. Even when differences in subsequent images are only caused by motion of objects, the prediction may be suboptimal because:

- the size of the prediction block (16*16) is too large;
- 3-D translations and rotations occur;
- fractional pixel displacements may occur;
- the search window may be too small;
- the search criterion may be suboptimal, for example, in case of lighting effects;
- the image in the frame memory contains quantization noise.

New video coding techniques, with better underlying source models, have recently appeared in literature. These techniques have been classified in Table 4-3. Work is underway in the ITU and ISO MPEG4 organizations to develop a long term video coding standard using technology of this type.

H.DLP - Data Interface

An optional data channel will be included to be multiplexed with the audio and video signals. Provision for high resolution still images using JPEG standard will be provided. It is a goal to interwork with other related ITU Recommendations. H.320 terminals and some existing videophone products. Table 4-4 is an example of a bitrate budget for the various virtual channels.

TABLE 4-3
Classification of Coding Techniques
Based on Source Models

LEVEL	SOURCE MODEL	CODED INFORMATION	CODING TECHNIQUE
1	Pels	Color of pels	PCM
2	Statistically dependent pels	Color of pels or block of pels	predictive coding transform coding
3	Translationally moving blocks	Color of blocks and motion vectors	Motion compensated hybrid DPCM/DCT coding
4	Moving structures	Mapping parameters or shape and motion	Fractal coding contour/texture coding
5	Moving unknown objects	Shape, motion and color of each object	Analysis/synthesis coding
6	Moving known object	Shape, motion and color of the known object	Knowledge based coding
7	Facial expressions	Action units	Semantic coding

H.22P - Multiplex/Error Control

It is proposed to employ packetized network and link layers to adaptively multiplex several virtual circuits -- speech, video, data, and supervision. This provides a greater degree of flexibility, and evolutionary growth potential, than TheSpeech Coder

The near term speech coder (AV.25Y) is to achieve as near toll quality as possible given the bit-rate budget. The long-term speech coder is expected to achieve toll quality at 4kbps. The long-term work has been referred to the speech experts within Working Party 15/2.

Schedule

The ITU has assigned an "urgent priority to the PSTN Videophone program, and the schedule for the near term recommendations is outlined below.

WP 15/1 Meeting Date	Target Status of the Near Term Recommendations
March '94	Initial Draft Completed
November '94	Draft Stabilized/Frozen
February '95	SG15 Applies for Resolution 1 Approval
November '95	ITU distributes Draft Recommendation for Approval by ballot

TABLE 4-4
Example of a Bitrate Budget for Very Low Bitrate Visual Telephony (kbit/s)

		Virtual Channel			
		Overhead/ Supervision (5%)	Speech	Video	Data
Overall Trans- mission Bit Rate	9.6 Kbps	0.5 Kbps	4.8 Kbps	4.3 Kbps[2]	Variable
			6.8	2.3[2]	
	14.4	0.7	4.8	8.9	Variable
			6.8	6.9	
	[3] ⋮	⋮	⋮	⋮	
	21.6	1.1	4.8	16.7	Variable
			6.8	14.7	
	[3] ⋮	⋮	⋮	⋮	
	28.8	1.4	4.8	22.6	Variable
			6.8	20.6	
Virtual Channel Bitrate Characteristic		Variable Bitrate	Dedicated, Fixed Bitrate [1]	Variable Bitrate	Variable Bitrate
Priority		High Priority	High Priority	Lowest Priority	Higher Than Video, Lower Than Overhead/ Speech

(1) The plan includes consideration of advanced speech codec technology such as... (a) a dual bitrate speech codec (b) reduced bitrate when voiced speech is not present.

(2) Achievement of motion video is considered to be questionable at 4.3 Kbps and inadequate at 2.3 Kbps.

(3) V.FAST operates at increments of 2.4 Kbps; i.e. 16.8, 19.2, 21.6, 24.0, 26.4, 28.8 Kbps.

(4) The channel priorities will not be standardized; the priorities indicated are examples.

4.4 IMPLICATIONS FOR THE GOVERNMENT

It is concluded from Section 4.3 that a series of ITU Recommendations (H.32P) will be developed in the near term for videophone and workstation interaction via the PSTN. It is likely that these Recommendations will have a major impact on the telecommunications structure within the Government community for the following reasons.

- At the current time, the presence of ISDN in the government is very restricted, and the timetable for its introduction is very speculative. These is no question that ISDN will one day be available in most government installations. The only question is; When? The PSTN is ubiquitous and pervasive today and will continue to be.

- Regardless of the ISDN timetable which will evolve, there will be some government installations in remote locations which will not receive ISDN for many years.

- In many national emergency scenarios, the PSTN would probably be more available than ISDN.

- The V.FAST modem standard, which will be finalized shortly, provides a high transmission bit rate -- approximately 25 Kbps.

- The PSTN videophone could provide an entry into the video phone service from which a user could migrate to the more capable ISDN video phone service.

- The PSTN videophone and ISDN video phone will be interoperable.

- The development of the international facsimile standard by the ITU created a quantum leap in the use of the PSTN for wide variety of services which was very difficult to anticipate or predict. The development of Recommendation H.32P could have a similar impact.

- Regional standards projects for the PSTN videophone are underway. The ITU Near-Term Project makes it possible for an International Standard rather than a Regional one.

+ ETSI has a project to develop a Draft Standard for a PSTN Videophone in late 1995.

+ TIAI.5 has initiated a project to develop an ANSI standard for the PSTN videophone.

5.0 VIDEO TRANSPORT VIA MOBILE RADIO

5.1 Mobile Radio Overview

5.1.1 Analog Cellular Radio

AMPS (Advanced Mobile Phone System) is the current air interface standard for the analog cellular system in the United States. At the time of this writing, the system is dominant throughout North America and has achieved a moderate amount of penetration on other continents as well. With an installed worldwide subscriber base in excess of 10 million units, it is highly probable that AMPS will survive past year 2000. A newer AMPS-like specification known as NAMPS has the potential to double, or triple, the number of analog subscribers in North America and provide some additional features now common to most digital cellular standards.[1][2][3]

The AMPS air interface is currently specified by the American National Standards Institute, Electronic Industries Association (EIA), and Telecommunications Industry Association (TIA). The current version is known simply as EIA/TIA-553. The base station transmit and receive bands are separated by 45 Mhz. The channel spacing is 30 Khz, and each operator within a geographical area is allocated exactly half of the available channels (416) for control and voice. The modulation technique for the user voice is frequency modulation.

Narrowband AMPS (NAMPS) is the name given to a more recent air interface compatibility specification recently sanctioned by the TIA. The current interim specification is divided into three parts: IS-88 (air interface), IS-89 (base station requirements), and IS-90 (mobile station requirements). NAMPS is a direct descendent of Narrowband Total Access communication System (NTACS), which is now enormously popular in Japan.

NAMPS takes each 30-Khz AMPS channel and splits it into three 10-Khz channels. Each cellular call is allocated a 10-Khz channel in NAMPS just as call is allocated a 30-Khz channel in AMPS. The resulting three-for-one split results in an increase in system capacity without the overhead of cell splitting and all its attendant headaches. NAMPS is compatible with the AMPS system in that the 30-Khz control channel is still used and mobile stations can be built to handle both standards beyond increased capacity which makes it attractive.

In the near term, AMPS will remain a viable cellular technology, offering enhancement features of interswitch or intermanufacturer handoff (IS-41) and

roaming. However, limitations in capacity and over-the-air call features (e.g., ISDN) requiring greater bandwidth will hamper AMPS from competing with future digital technologies.

5.1.2 Digital Cellular Radio

Digital cellular technology provides solutions to the capacity problem and other issues prevalent in today's analog cellular systems. Improvements in the areas of mobility, cost, power consumption, and spectrum utilization are provided by conversion to digital as listed below.

- **Increased radio frequency spectrum efficiency and system capabilities**
  Digital multiple access techniques (MATs) are being developed to provide additional capacity within the current spectrum allocations for cellular communications. Analog systems cannot provide these advanced capabilities.

- **A signal that can be almost free of noise and interference impairments**
  Digital techniques support error correction so that most static, fading, and other disturbances to the radio channel are undetectable to the customer or negligible compared to today's analog noise problems.

- **Low power consumption by the system components**
  Digital components use less power than analog components.

- **Improved security features and services**
  Because it is easier to encrypt a digital signal than an analog signal, customers will benefit from higher levels of security services.

Digital cellular is in the developmental stage in the U.S. Key issues remaining to be resolved are listed below.

- Ensuring interoperability and compatible interfaces to current analog systems during the transition to a completely digital environment. This may be accomplished via a dual mode system in the transition period.

- Providing customers complete mobility to communicate anywhere with

cellular phones, pagers, pocket phones, and wireless Private Branch Exchanges (PBXs). This capability heralds widespread use of the Personal Communications Services (PCS).

- Implementing network architectures that eliminate current multipath fading environments and ensure soft (no-break) hand-offs.

- Developing a seamless internetwork architecture that allows users to roam nationwide or even worldwide. Such an architecture implies implementation of an Intelligent Network (IN) that has a common data base and provides customer identification and service as the user travels about the seamless environment.

- Satisfying customer demand for improved security services. The authentication and additional security services provided by digital systems will minimize fraud and other security problems common today.

- Agreeing on a standard multiple access technique. On January 6, 1992, the Board of Directors of the Cellular Telecommunications Industry Association (CTIA) unanimously passed a resolution stating that time division multiple access (TDMA) should remain the industry standard for digital. CTIA is the primary association promoting digital cellular standards in the United States. TDMA and two other advanced multiple access techniques are discussed in more detail below.

Time Division Multiple Access (TDMA)

In the proposed TDMA system, each communications channel, which is 30 kilohertz (Khz) wide, is shared by several users. TDMA divides a channel into six time slots and assigns two time slots to each user. By so doing, TDMA allows three users to share the channel bandwidth. CTIA has proposed a 40-millisecond frame to be shared equally among the three users of the same channel. Applying this time period to TDMA represents a fivefold increase in channel sharing capacity over the current analog FDMA system.

Code Division Multiple Access (CDMA)

CDMA, the second MAT being proposed within the digital cellular industry, employs wideband transmission techniques to achieve capacity gains. CDMA, also known as spread spectrum, was developed for military applications just after World War II. In CDMA transmission, each user's baseband signal is dispersed over the entire 1.25 Mhz channel. CDMA's system gain is due to the sharing of the same 1.25 Mhz channel by many users, each at a low power level. Each signal's power is spread over the whole 1.25 Mhz bandwidth so that each user's contribution of energy per hertz is small.

Extended Time Division Multiple Access (E-TDMA)

E-TDMA, the third MAT being proposed by the industry, is being developed by Hughes Network systems (HNS) for cellular applications. The cellular industry, however, is not considering E-TDMA for first generation digital systems.

HNS is incorporating digital speech interpolation (DSI) into E-TDMA to gain additional capacity. DSI takes advantage of the fact that the average channel utilization for speech is less than 50 percent. In DSI, a user occupies a channel only when the user is active. When the user pauses, the channel can be reassigned to another user. By having a large number of users sharing a small number of channels, contention often occurs. However, the collisions will be brief.

5.1.2.1 Digital Cellular Standards

The need for increased system capacity for current cellular systems drives the development of digital cellular standards worldwide. Europe, Japan, and the United States are developing digital cellular standards for their respective national cellular communities. Table 5-1 displays the technical characteristics of the developing digital cellular standards in North America, Europe, and Japan. These efforts are not being coordinated with each other, resulting in three distinct digital cellular standards profiles. Since the three competing cellular standards use different technical properties the networks are incompatible. Due to the incompatibility of the three standards, a truly interoperable international cellular network is currently impossible. However, further development of new digital cellular standards may eventually provide worldwide seamless roaming to any cellular user.

TABLE 5-1

Technical Characteristics of Digital Cellular Standards

	GSM	ADC	JDC
Proponent	Europe	United States	Japan
Multiple Access Technique	TDMA	TDMA	TDMA
Carrier spacing	200 Khz	30 Khz	25 Khz
Users (channels) per carrier	8	3	3
Voice bit rate	13 kb/s	8 kb/s	8 kb/s
Carrier bit rate	270 kb/s	48.6kbs	42kb/s
Diversity methods	Frequency hopping		Antenna diversity
Bandwidth per voice channel	25kbz	10 Khz	8.3 Khz
Required C/I	9 Db	16 Db	13 Db

ADC: American Digital Cellular
C/I: Carrier-to-Interface Ratio
GSM: Global System for Mobile Communications
JDC: Japanese Digital Cellular

GSM (EUROPE)

The digital cellular standard closest to full-scale implementation exists in Europe. Through the European Telecommunications Standards Institute, a group of European nations developed a digital cellular standard known as Groupe Speciale Mobile (GSM). The GSM standards use a TDMA technique for efficient use of the cellular spectrum. GSM cellular has been allocated new spectrum in Europe, separate from the current analog frequencies. The new frequencies were allocated with the cooperation of the frequency assignment administrations of all participating nations. More than 20 European countries have signed a memorandum of understanding that calls for GSM service on the main transport

routes between the European capital cities by 1995.

(ADC) UNITED STATES

The United States is adopting a dual-mode standard, which allows analog terminals to continue using the current frequencies, and still allows for the improvement in channel capacity by using a digital signaling technique. The TDMA standard being developed in the United States is often referred to as American Digital Cellular (ADC).

The cellular radio community in the United States produces standards in a subcommittee sponsored by the Telecommunications Industry Association (TIA) and the CTIA (Cellular Telecommunications Industry Association). TIA and CTIA have created the Cellular Standards committee, TR45, to deal with cellular issues. There are five subcommittees that perform the standards work:

- TR45.1 - Analog Cellular
- TR45.2 - Cellular Intersystem Operations
- TR45.3 - Digital Cellular
- TR45.4 - Personal Communications Services
- TR45.5 - Wideband Spread-Spectrum Digital Technology

The current ANSI-accredited standard for analog cellular radio is Electronic Industries Association/Telecommunications Industry Association (EIA/TIA) 553, *Mobile Station-Land Station Compatibility Standard*. The most active subcommittees, TR45.2 and TR45.3, are developing Interim Standards to support the new dual-mode cellular system that is being planned for the transition to all digital services. The TR45.5 subcommittee was established in early 1992 to investigate a wideband MAT for digital cellular application. The TR45.1 subcommittee is pursuing a course to standardize the N-AMPS in the United States.

The TIA has recently completed the interim standard designated as IS54 which fully defines the digital cellular system for TDMA operation. By employing 3 time slots a voice channel has a gross bitrate of 13Kbps and a net payload of 8Kbps. Deployment of operational equipment based on this standard began in 1993. The TIA has also completed an interim standard based on CDMA technology which is designated as IS96.

5.1.3 Personal Communication Services

The Telecommunications Industry Association (TIA) has defined Personal Communications Services (PCS), or wireless communications, as "a mobile radio voice and data service for the provision of unit-to-unit communications which is based on microcell and other technologies that enhance spectrum capacity to the point where it will offer potential for essentially ubiquitous and unlimited, untethered communications." This ambitious definition is far more wide-reaching than the current wireless reality. Most wireless telecommunications systems cannot provide reliable communication as distance becomes a factor, nor is the technology as universal as the TIA definition implies.

Experts estimate that PCS will have 115 million subscribers by the year 2000. Varying user requirements will lead to a wide range of access methods. Telecommunications industry experts believe that PCS could provide a variety of services instead of merely acting as a pocket-size digital cellular telephone. The services could include wireline and wireless services on either a local or long-distance basis for the home or office.

PCS has become the catch-all term for mobile, switched voice services. Proposed PCSs fall into two basic categories. Personal Communications Network (PCN) is similar to existing cellular systems, but the cell size may be as small as a 300-foot radius. PCN may be best for low-power, low-cost pocket handsets. The second category, cordless telephone-second generation (CT-2), is essentially a one-way call-out system. Its chief attraction is low cost.

5.1.4 UPT (Universal Personal Telecommunications)

UPT is a telecommunications service concept that provides for personal rather than terminal mobility. This service concept enables a person to initiate and receive calls on the basis of a unique, personal, network-transparent UPT number. Personal mobility is conferred by the user's ability to access a telecommunications service at any terminal. Users may configure terminals to meet their unique requirements, limited only by the terminal and network capabilities and restrictions imposed by the network provider. Personal mobility is limited only by the network's capability to locate users on the basis of unique UPT numbers used for addressing, routing, and charging the user's calls.

UPT service requires the transfer of control information between various databases to establish the communications environment for the call. Different call handling procedures are used according to whether calls are routed within a single

network or across multiple networks. The integrated network functions that are an integral part of the evolving network for UPT include access, transport, intelligence, and management.

Recent developments in emerging technologies such as Intelligent Network (IN), Integrated Services Digital Network (ISDN) and digital cellular communications are providing new dimensions to personal telecommunications capabilities. These capabilities make new services such as UPT possible. UPT represents a complex concept still in its infancy, with the pace of its evolution dependent upon market needs and advance in IN technology.

The ANSI T1 committee established the T1P1 committee in late 1990 to define and develop standards in the area of Personal Communications, which includes UPT. Within the T1P1 committee are three working groups responsible for the following activities:

- T1P1.1 has responsibility for program management within the development of personal communications standards.
- T1P1.2 has responsibility for functional requirements, interfaces and the overall PCS architecture.
- T1P1.3 has responsibility for service descriptions, functional models and network interfaces.

Provision of UPT will occur gradually, beginning with a simplified set of UPT features and capabilities, such as voice only, that later progresses into more advanced scenarios that include voice, data, and imagery.

The UPT service is described as universal for the following reasons:

- UPT operates on any terminal on any network.
- UPT supports all basic telecommunications services (e.g., telephone, mobile, facsimile, video).
- UPT is available globally (from any geographic location).

UPT service is described as personal for the following reasons:

- The user has a UPT number that is associated with the person rather than with a terminal.

- Personal mobility (i.e., mobility between terminals) is provided to the user.

- UPT enables personalization of various service features.

UPT is a service concept, independent of the network, that takes advantage of the new developments in telecommunications such as the ISDN, Future Public Land Mobile Telecommunications Systems (FPLMTS), and the mobile satellite systems which are entering the telecommunications world in this decade. Some of these concepts are currently being tested, including second and third generation cordless telephone (CT2/CT3) and other experimental wireless services.

5.1.5 Future Public Land Mobile Telecommunications Systems (FPLMTS)

FPLMTS are systems that will provide telecommunications services to mobile or stationary users by means of one or more radio links. This mobility will be unrestricted in terms of location within the radio coverage area. FPLMTS will be an integral part of the Public Switched Telephone Network (PSTN) and will extend the telecommunications services of fixed networks to mobile or stationary users over wide geographic areas. The constraints on the system will be those imposed by spectrum allocation and radio propagation. FPLMTS will allow users to originate and terminate calls from small, lightweight portable devices, regardless of location. FPLMTS will provide voice grade services for making and receiving calls from anywhere, and should be simple to use. The security and quality of FPLMTS will be comparable to that of wireline services. From a user's perspective, the access device will probably be a small, shirt-pocket-sized portable terminal. FPLMTS will have low-power radio access to cellular sites that are connected to the PSTN. FPLMTS work is being performed by the Radiocommunication Section of the standardization ITU. A summary of the key features of FPLMTS are:

*high degree of commonality of design worldwide.
*compatibility of services within FPLMTS and with the fixed networks.
*high quality.
*use of a small pocket terminal worldwide.

Appendix 5.1 contains a summary of the FPLMTS project. Appendix 5.2 describes the relationship between FPLMTS and UPT.

TABLE 5-2 FLPMTS Environments

ENVIRONMENT				
CLASS	A	B	C	
Business Indoor	*			A
Neighborhood Indoor/outdoor		*		B
Home	*			A
Urban vehicular outdoor		*		B
Urban pedestrian outdoor	*			A
Rural outdoor		*		B
Fixed outdoor		*		B
Urban satellite		*	*	C&B
Rural satellite			*	C
Fixed satellite			*	C
Indoor satellite			*	C

TABLE 5-3 Speech Coder Parameters

PARAMETER							
CLASS	A		B		C		
	Req	Obj	Req	Obj	Req	Obj	Units
Speech quality without errors	G726		G726		G726		
Quality loss, two radio interfaces and transcoding							MOS
Coder/decoder delay (one way)	5	2	20	10	100	40	ms
Power consumption	2	2	20	5	300	200	mw
Speech coder bit rate	32	16	16	4	4	2-3	kb/s
Gross speech bit rate	40	24	32	8	8	4	kb/s
Adaptive bit rate	no	no	yes	yes	no	no	
Voice activity detection	no	no	no	yes	yes	yes	
Transparent to DTMF	no	yes	no	yes	no	no	

5.1.6 Mobile Satellite Systems

Mobile satellite systems are currently being investigated for use with cellular and hand-held terminals. The recent World Administrative Radio Conference (WARC) in Barcelona allocated small amounts of spectrum in the L-band (1.5 Ghz) and S-band (2.5 Ghz) for satellite applications. currently, a number of different satellite architectures are being discussed:

- Motorola has proposed a system called Iridium, a 66 satellite low-earth orbit (LEO) system, which would provide worldwide communications. These LEO satellites would be in orbit approximately 500 miles above the earth. The Iridium system would use a lightweight, hand-held terminal with a stub antenna.

- Loral Space and Qualcomm are proposing the Globalstar system, which has 48 satellites in LEO (orbiting at about 800 miles above the

earth). Subscribers would receive direct satellite services through hand-held terminals.

- TRW is proposing a system called Odyssey, consisting of 9 satellites at mid-earth orbit (MEO). These satellites would orbit approximately 6000 miles above the earth. The system plans to use the same satellite bands as the previous systems and may include Ku-band. Odyssey utilizes a phased-array stub antenna to access the satellites.

5.2 VTC Application of Mobile Radio

At a meeting on 6-17 September 1993, Working Party 15/1 re-appointed a Rapporteur for Very Low Bitrate Videophone. Guidance given to the Rapporteur by the ITU is provided in Appendix 5.3. One of the requirements is to develop a Recommendation for operation over mobile radio. Several contributions have been presented to the LBC Group on this topic, four of which are included in the appendices listed below.

Appendix	Title
5.4	Video Coding in Mobile Networks - Some Aspects
5.5	Proposal for a Generic Data Stream Structure Considering Current Short-term and Long-term Standardization Activities
5.6	R2072: MATV - The Mobile Audio-Visual Terminal
5.7	R2072: Mobile Audio-Visual Terminal (MAVT): Liaison Statement to TSS Study Group 15, Special Rapporteurs Group on very low bitrate video coding (Schaphorst Group)

Highlights of these appendices are provided below.

5.4 - Video Coding in Mobile Network

This document points out some important aspects of mobile networks (European Universal Mobile Telecommunication Systems. FPLMTS) and gives guidelines for the development of video coding algorithms for the mobile application. It is recommended that the algorithms which will be developed by

MPEG-4 be strongly considered.

5.5 - Generic Data Stream

In this document, a proposal for a generic data stream structure is described. A common and flexible data stream structure is desireable because there will be a fast development of services and demands in the area of mobile telecommunications. It is difficult to fix a specific data structure and simultaneously fulfill future demands of compatibility with existing devices. A flexible data stream structure can handle a broad variety of applications.

5.6 - The European Mobile Audio-Visual Terminal

This document is an overview summary of the European RACE project R2072 which is designated "The Mobile Audio-Visual Terminal" (MAVT). The project objective is to find powerful video and audio coding algorithms for the transmission of moving and still video in a mobile environment, and implement them on a demonstrator. This necessitates a study of user requirements, network and channel-characteristics, service definitions and a general terminal architecture. The project will deliver future algorithms for low bitrate video (px 8k bit/s) and audio coding, and a futuristic terminal with several novel features, including a variable resolution display and good quality audio. New proposals for VLSI chips for video and audio coding are expected from realisation of the demonstrator.

5.7 - R2072 Mobile Audio Visual Terminal: Channel Coding Aspects

The main technical objectives of the MAVT project are the development of robust video and audio coding algorithms for transmission of moving and still video and associated audio in mobile networks. Due to a variety of channel impairments encountered in case of a wireless mobile propagation medium special counter-measures have to be undertaken to guarantee the target transmission quality. This document discusses error control strategies under investigation in the context of European MAVT project.

5.3 Government Applications

At the present time the vast majority of operational cellular radio systems throughout the world are based on analog technology. Work is underway to convert the systems to a digital format, but this will require some time to accomplish. Since digital operation is a prerequisite for video telephony, audiovisual services are not available in the mobile radio environment today. As work on U.S. digital cellular unfolds the primary payload bitrate for a voice channel is 8kbps. This low bit rate relative to that available on the PSTN (14.4 Kbps, 28 Kbps) reflects the fact that the mobile radio link is more complex and fragile than the PSTN.

Videotelephony work which has been accomplished in the PSTN suggests that it is very difficult to achieve satisfactory videophone performance around 8Kbps. Consequently it is likely that channel aggregation techniques will have to be employed. If two voice channels are multiplexed to provide a combined bit rate of 16Kbps it would be possible to achieve satisfactory videotelephony performance as outlined in section 4.0 of this report. In Europe the term Px8Kbps has been coined to represent this multichannel concept.

When the mobile radio infrastructure evolves into a digital form, videotelephony services will naturally evolve. There are many applications in the Government community which could advantageously employ such services some of which are listed below.

- There are many instances where it is desirable to transmit video from a remote field location where no phone exists.

- There would be instances where it would be desirable to access video from a database at a remote mobile location. The video could be used to assist in a troubleshooting mission.

- The mobile location could be in a moving platform (aircraft, truck, ship) as well as land based.

- This mobile visual capability is particularly well suited for use in disaster scenarios where agencies like FEMA perform critical missions.

REFERENCES

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[2] Electronic Industries Association, "EIA/TIA-553 Mobile Station - Land Station Compatibility Specification," Telecommunications Industry Association, 1989.

[3] Boucher, J. R.., *The Cellular Radio Handbook: A Reference for Cellular System Operation*, Quantum Publishing, 1990.

6.0 VIDEO TRANSPORT VIA INTERNET

6.1 Internet Overview

Internet refers to a unique collection of networks, mainly, in the U.S., but also throughout the world, most of which are built using the Transmission Control Protocol/Internet Protocol (TCP/IP) protocol suite and all of which share a common name and address space. Computers on the Internet use compatible communications standards and share the ability to contact each other. Users of the Internet communicate mainly via electronic mail (e-mail), via Telnet, a process that allows them to login to a remote host, and via the File Transfer Protocol (FTP), a protocol that allows them to transfer information on a remote host of their local site. See Figure 6.1.

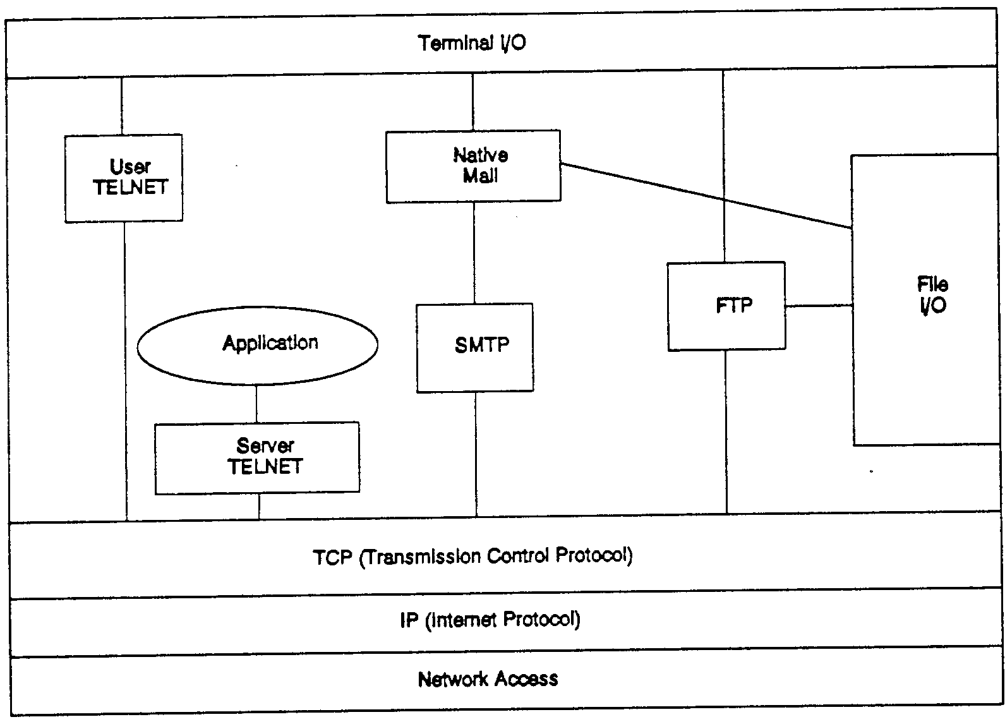


Figure 6.1 Conceptual Structure for Internet Protocol

Currently, there are over 100 countries that have some access to the Internet, almost 39,000 networks assigned unique IP network numbers, almost one million hosts known to the domain name system, and at least 3,500,000 users worldwide.

The Internet exists to facilitate the sharing of resources among participating organizations, which include government agencies, educational institutions, and

private corporations; to promote collaboration among researchers; and to provide a testbed for new developments in networking.

The Internet is a cooperating group of independently administered networks. Each component network on the Internet has its own administrative body, its own policies, and its own procedures and rules. There is no central, overseeing authority for the whole of the Internet. However, certain government agencies have traditionally been prominent in setting policy that is followed throughout the Internet. Today important policy decisions come from the National Science foundation (NSF), administrator of the National Science Foundation Network (NSFNET), and from the Defense Information Systems Agency (DISA), administrator of the Defense Data Network (DDN).

In addition to influential government agencies, a strong, mostly voluntary, coalition of technically knowledgeable individuals guide the development of the Internet. Innovations that upgrade the technical quality of the Internet are usually worked out cooperatively by the technical community working together under the auspices of a group called the Internet Architecture Board (IAB). The IAB has recently been included as part of the structure of the Internet Society (ISOC). There is a hierarchy of Task Forces, Areas, and Working Groups under the IAB that address technical problems and develop solutions. Eventually, solutions are agreed upon by the Internet community and the IAB recommends they be implemented. In this way, new additions to the TCP/IP suite of protocol standards are developed, tested, recommended, and implemented.

ARPANET

The ARPANET does not exist any more, but we include its description because you will sometimes hear the term ARPANET used interchangeably with the term Internet. The ARPANET, the predecessor of today's Internet, was the first packet-switched network to connect heterogeneous computers. That is, computers of different types could exchange information for the first time because of the network protocols developed for the ARPANET.

In 1984, the ARPANET was split into two networks: the ARPANET for research oriented activities, and the Defense Data Network (DDN) for military operational activities. The DDN still exists as one of the Internet networks; its MILNET network provides unclassified operational support to military users. The ARPANET itself was phased out in 1990 in favor of the more advanced NSFNET backbone.

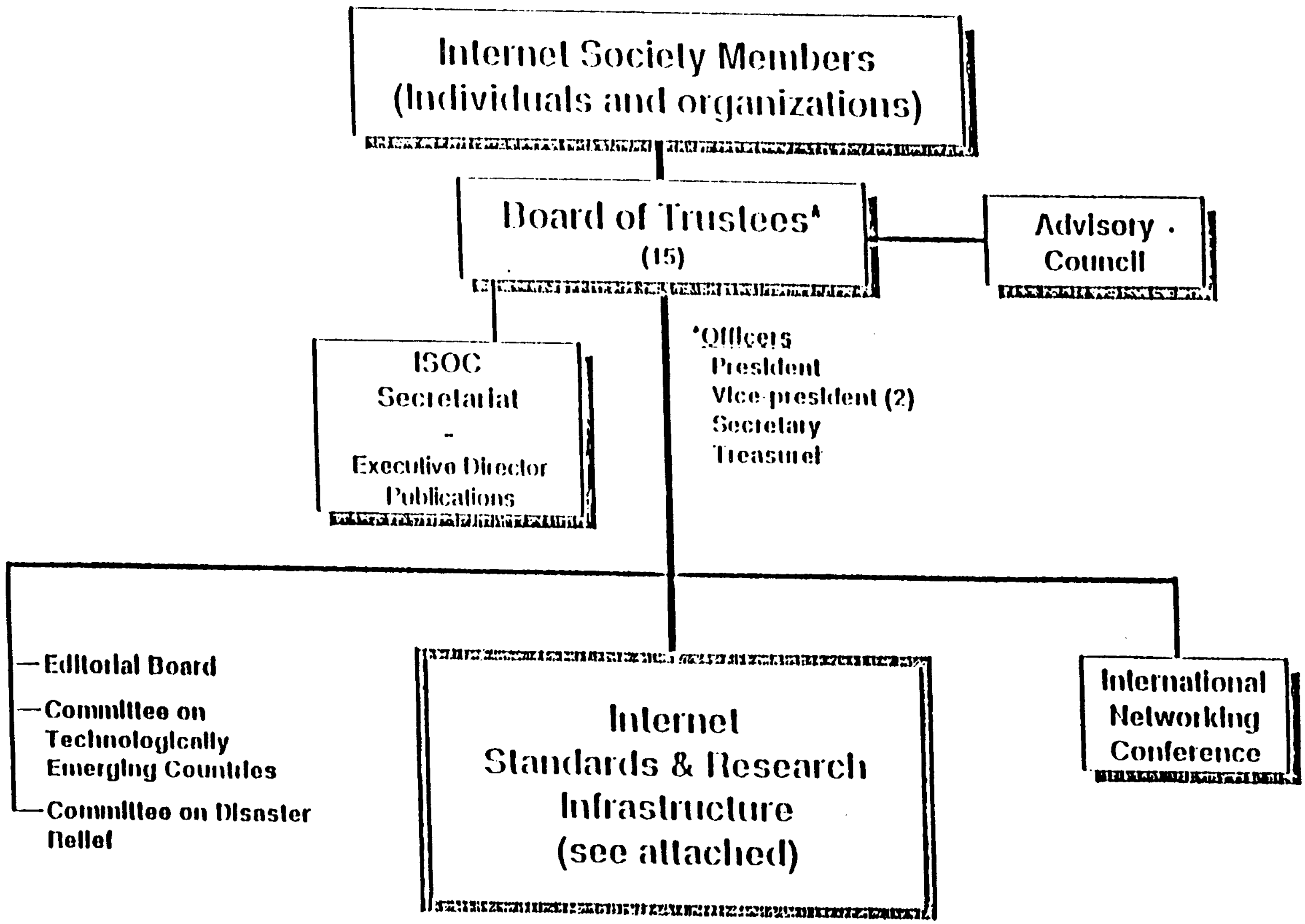
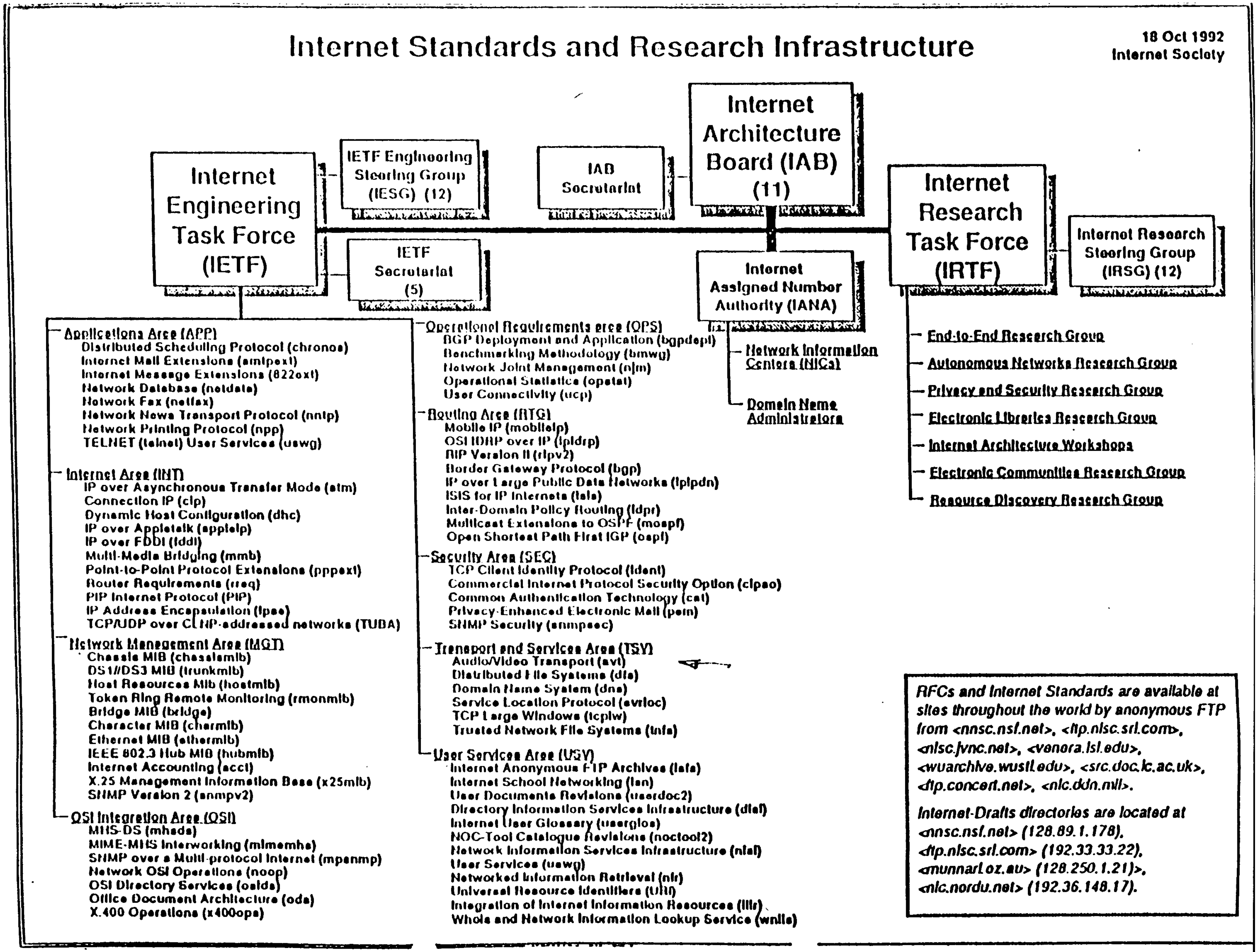


FIGURE 6.2
INTERNET SOCIETY MEMBERS



Internet Standards and Research Infrastructure
18 Oct 1992
Internet Society
Internet Engineering Task Force (IETF)
IETF Engineering Steering Group (IESG) (12)
IETF Secretariat (5)
IAB Secretariat
Internet Architecture Board (IAB) (11)
Internet Assigned Number Authority (IANA)
Internet Research Task Force (IRTF)
Internet Research Steering Group (IRSG) (12)
Applications Area (APP)
Distributed Scheduling Protocol (chronos)
Internet Mail Extensions (smtpext)
Internet Message Extensions (822ext)
Network Database (netdata)
Network Fax (netfax)
Network News Transport Protocol (nntp)
Network Printing Protocol (npp)
TELNET (telnet) User Services (uswg)
Internet Area (INT)
IP over Asynchronous Transfer Mode (atm)
Connection IP (cip)
Dynamic Host Configuration (dhc)
IP over Appletalk (appleip)
IP over FDDI (fddi)
Multi-Media Bridging (mmb)
Point-to-Point Protocol Extensions (pppext)
Router Requirements (rreq)
PIP Internet Protocol (PIP)
IP Address Encapsulation (ipae)
TCP/UDP over CLNP-addressed networks (TUBA)
Network Management Area (MGT)
Chassis MIB (chassismib)
DS1/DS3 MIB (trunkmib)
Host Resources Mib (hostmib)
Token Ring Remote Monitoring (rmonmib)
Bridge MIB (bridge)
Character MIB (charmib)
Ethernet MIB (ethermib)
IEEE 802.3 Hub MIB (hubmib)
Internet Accounting (acct)
X.25 Management Information Base (x25mib)
SNMP Version 2 (snmpv2)
OSI Integration Area (OSI)
MHS-DS (mhsds)
MIME-MHS Interworking (mimemhs)
SNMP over a Multi-protocol Internet (mpsnmp)
Network OSI Operations (noop)
OSI Directory Services (osids)
Office Document Architecture (oda)
X.400 Operations (x400ops)
Operational Requirements area (OPS)
BGP Deployment and Application (bgpdepl)
Benchmarking Methodology (bmwg)
Network Joint Management (njm)
Operational Statistics (opstat)
User Connectivity (ucp)
Routing Area (RTG)
Mobile IP (mobileip)
OSI IDRP over IP (ipidrp)
RIP Version II (ripv2)
Border Gateway Protocol (bgp)
IP over Large Public Data Networks (iplpdn)
ISIS for IP Internets (isis)
Inter-Domain Policy Routing (idpr)
Multicast Extensions to OSPF (mospf)
Open Shortest Path First IGP (ospf)
Security Area (SEC)
TCP Client Identity Protocol (ident)
Commercial Internet Protocol Security Option (cipso)
Common Authentication Technology (cat)
Privacy-Enhanced Electronic Mail (pem)
SNMP Security (snmpsec)
Transport and Services Area (TSV)
Audio/Video Transport (avt)
Distributed File Systems (dfs)
Domain Name System (dns)
Service Location Protocol (svrloc)
TCP Large Windows (tcplw)
Trusted Network File Systems (tnfs)
User Services Area (USV)
Internet Anonymous FTP Archives (iafa)
Internet School Networking (isn)
User Documents Revisions (userdoc2)
Directory Information Services Infrastructure (disi)
Internet User Glossary (userglos)
NOC-Tool Catalogue Revisions (noctool2)
Network Information Services Infrastructure (nisi)
User Services (uswg)
Networked Information Retrieval (nir)
Universal Resource Identifiers (URI)
Integration of Internet Information Resources (iiir)
Whois and Network Information Lookup Service (wnils)
Network Information Centers (NICs)
Domain Name Administrators
End-to-End Research Group
Autonomous Networks Research Group
Privacy and Security Research Group
Electronic Libraries Research Group
Internet Architecture Workshops
Electronic Communities Research Group
Resource Discovery Research Group
RFCs and Internet Standards are available at sites throughout the world by anonymous FTP from <nnsc.nsf.net>, <ftp.nisc.sri.com>, <nisc.jvnc.net>, <venera.isi.edu>, <wuarchive.wustl.edu>, <src.doc.ic.ac.uk>, <ftp.concert.net>, <nic.ddn.mil>.
Internet-Drafts directories are located at <nnsc.nsf.net> (128.89.1.178), <ftp.nisc.sri.com> (192.33.33.22), <munnari.oz.au> (128.250.1.21)>, <nic.nordu.net> (192.36.148.17).

FIGURE 6.3

NSFNET

The National Science Foundation (NSF) sponsors the main research and education backbone of the Internet today. This network is called the NSFNET. The NSFNET is a hierarchical network of networks. At the highest level is the backbone. At first , the NSFNET backbone connected supercomputer centers, but the NSFNET evolved to interconnect a group of networks commonly called "mid-level" networks.

MILNET

The MILNET was established as a separate network in 1984 when it was partitioned off of the ARPANET. This network is supported by the Defense Information Systems Agency (DISA) of the Department of Defense. The MILNET is the unclassified component of the Defense Data Network (DDN). There are currently six gateways, called "mailbridges," that link the MILNET with the NSFNET/Internet.

INTERNET ORGANIZATION

Figure 6.2 is an organization chart of the Internet Society showing its relationship to the Internet Standards and Research Infrastructure which is illustrated in Figure 6.3.

Experimented work in the transmission of audio and video over Internet is being performed within the Transport and Service Area (TSV). Within the TSV are the two related Work Groups listed below.

Audio/Video Transport (AVT)

The Audio/Video Transport Working Group, chaired by Stephen Casner, USC, was formed to specify experimental protocols for real-time transmission of audio and video over UDP (User Datagram Protocol) and IP multicast. The focus of this Group is near-term and its purpose is to integrate and coordinate the current AV transport efforts of existing research activities. No standards-track protocols are expected to be produced because UDP transmission audio and video is only sufficient for small-scale experiments over fast portions of the Internet. However, the transport protocols produced by this Working Group should be useful on a larger scale in the future in conjunction with additional protocols to access

network-level resource management mechanisms. Those mechanisms, research efforts now, will provide low-delay service and guard against unfair consumption of bandwidth by audio/video traffic.

The AVT Working Group may design independent protocols specific to each medium, or a common, lightweight, real-time transport protocol may be extracted. Sequencing of packets and synchronization among streams are important functions, so one issue is the form of timestamps and/or sequence numbers to be used. The Working Group will not focus on compression or coding algorithms which are domain of higher layers.

Multiparty Multimedia Session Control (MMUSIC)

The purpose of the MMUSIC Working Group is to develop an infrastructure to support teleconferencing over Internet. Multimedia session control, defined as the management and coordination of multiple sessions and their multiple users in multiple media (e.g., audio,video), is one component of the infrastructure. The Multiparty Multimedia Session Control Working group is chartered to design and specify a protocol to perform these functions.

The protocol will provide negotiation for session membership, underlying communication topology and media configuration. In particular, the protocol will support a user initiating a multimedia multiparty session with other users over the Internet by allowing a teleconferencing application on one workstation to explicitly rendezvous with teleconferencing applications running on remote workstations. Defining a standard protocol will enable session-level interoperability between different teleconferencing implementations.

The focus of the Working Group is to design a session negotiation protocol that is tailored to support tightly-controlled conferences. The current protocol carries primarily loosely-controlled sessions, i.e., sessions with little to no interaction among members and with no arbitration facility, security, or coordination of quality-of-service options for time-critical media.

The main goal of the Working Group will be to specify the session control protocol for use within teleconferencing software over the Internet. The Working Group will focus on the aspects of the session control problem that are well understood, while keeping an eye on evolving research issues. Toward this end, the Working Group has made an inventory of existing conferencing systems and their session control protocols. The Working Group will document the requirements of the existing prototypes as a basis for the protocol development. The Working Group will iteratively refine the protocol based on implementation and operational

experience.

6.2 Video Coding Products

Teleconferencing experiments on Internet have been performed using three different video coding software packages installed on Workstation platforms. The software packages are briefly described below.

PICWIN by Bolt Beranek and Newman

PictureWindow (PICWIN) is a software package that allows workstation users to hold video conferences over IP networks. BBN's PictureWindow system brings multiparty, wide-area video conferences directly onto your workstation screen with minimal additional hardware. PictureWindow provides personal video conferencing capability through the use of BBN software, Sun's inexpensive VideoPix frame-capture board, and a video camera on your existing color SPARCstation.

PictureWindow compresses and decompresses video entirely in software, transmits it using standard IP protocols, and displays it in multiple windows under OpenWindows or the XWindow System. It operates over Internet Protocol (IP) based wide-area networks, and it can be used in either point-to-point or multicast modes.

PictureWindow video windows measure 320x240 pixels with 16 levels of gray. Refresh rate depends upon system and network load, but it is typically between 3 and 6 frames per second.

PictureWindow sends compressed video in UDP/IP datagrams and functions in both local and wide-area network environments. The actual network bandwidth used by any one conferee depends on the amount of motion in the supplied video, and the image quality desired by the viewers. In two-way conferences, each conferee can selectively adjust the equality and compression parameters for the image they are viewing. PictureWindow functions best when network paths with at least 256 kilobits/second are available. Nevertheless, network paths as slow as 56 kilobits/second can be used by decreasing the frame rate and increasing the acceptable image error. More information on PICWIN is provided in Appendix 6.1.

NV by Xerox

The NV software package encodes frame differences using wavelet coding over an 8 x 8 pixel array. Motion compensation is not employed. At the default bitrate of 128 Kbps the picture rate is 3 frames/second. Picture resolution is 320 x 240 pixels.

IVS by Inria

Inria is a research institute in France which has developed the IVS software package which is based on the H.261 coding algorithm. A picture frame rate of 1-2 frames/second has been achieved in experiments.

CUSEEME by Cornell University

Cornell University has developed a software video coding software package which is suitable for the Macintosh PC as well as the Sun Workstation.

6.3 Government Applications

The Internet is now used to provide E-Mail service to millions of users throughout the world. The primary use for Internet is the provision of a store-and-forward message service as opposed to an interactive conversational service (speech, video). Nevertheless, Internet is becoming so pervasive and ubiquitous that it is natural to consider the extension of Internet to provide videotelephony services.

Since E-Mail is a personalized service it is natural to consider the use of Internet for head-and-shoulders videophone applications as opposed to the videoconferencing application from a conference room. However, as desktop multimedia workstations become more prevalent and video bridges (Multiport Control Unit) become more capable, video conferences (like audio conferences today) will become more desk oriented rather than conference room oriented. Consequently, in time, Internet could be considered to provide video conferencing service as well as videophone service.

At the present time the use of Internet to provide videotelephony services must be considered very speculative. It may never have a serious impact as a communications medium relative to other media such as switched 56kbps, N-ISDN, PSTN, and LANS/WANS. Reasons for this viewpoint are listed below.

- Typical transmission bit rates employed for Internet are 128 Kbps for video and 64 Kbps for speech. The Internet infrastructure does not support these rates in high volume.

- The protocol structure is not well suited to low delay interactive services.

7.0 CONCLUSIONS AND RECOMMENDATIONS

Video teleconferencing is becoming a very important service throughout the government community. At the present time the traditional transport media are switched 56Kbps channels and fractional TI channels. In this report four alternative transport media for the VTC application have been examined, and an overview of the result is provided in Table 7-1.

TABLE 7-1
Overall Summary

<table>
<tr><th></th><th>TRANSPORT MEDIUM</th><th>VTC TERMINAL EQUIPMENT</th><th>GOVERNMENT APPLICATION</th></tr>
<tr><td>LAN/MAN</td><td>Several mature configurations exist: Ethernet, Token Ring, FDDI, ATM</td><td>Many VTC terminals are available; some meet H.261; H.32Z standardization is underway</td><td rowspan="3">VTC via these media are very important to the government; mobile radio is in the future</td></tr>
<tr><td>PSTN</td><td>Ubiquitous, mature; new modem technology increases the potential for VTC</td><td>Several proprietary products exist; ITU standard being developed</td></tr>
<tr><td>MOBILE RADIO</td><td>Not ready for VTC applications until conversion to digital operation gains momentum</td><td>No products exist; Europe is planning a VTC product; ITU standard is being developed</td></tr>
<tr><td>INTERNET</td><td>Used primarily for E-Mail. Existing Internet not well suited for high volume of VTC traffic</td><td>Only software products are employed; one is commercially available</td><td>This experimental work may be applicable to future WANs such as NREN</td></tr>
</table>

Since there is considerable promise for the use of these media for VTC

service in the government community, it is recommended that the government actively participate in the process of stimulating technology advancement and standards development in the following specific areas.

- H.32Z; conversion of H.32O to LAN operation

- H.32P; development of a videophone standard for the PSTN and mobile radio environment

- development of digital mobile radio standards

- further analysis of LAN/MAN technology and related VTC terminal products to provide guidance for government deployment.

APPENDIX 3.1

EXAMPLES OF COMMERCIAL PRODUCTS FOR PROVIDING VTC SERVICES OVER LANs

> *Note: Because PictureWindow uses the MIT-SHM option of X Windows, you MUST run PictureWindow on the same workstation as your X server. You cannot run picwin on a remote computer and use an X terminal or other X server to view the results.*

To conduct conferences with workstations other than your own, you will need to be connected to a TCP/IP network, and your workstation should be configured with a unique IP address on that network. Your system or network administrator should be able to help you if you are uncertain regarding this requirement. Most Sun workstations are connected to networks of some type (usually Ethernet) and are configured for TCP/IP operation as the default. You can verify your networking capabilities by consulting the *Sun System & Network Manager's Guide* supplied with your workstation or your local network administrator.

If you are setting up your workstation for the first time, a configuration using the Sun-supplied GENERIC configuration of SunOS 4.1.1, OpenWindows 2.0, and the VideoPix driver are sufficient to support PictureWindow.

VIS-A-VIS

VIS-A-VIS VIDEO

Full Motion Video on your PC

VIS-A-VIS Video is desktop conferencing software that delivers full motion video information through your PC. Since it runs on your PC it is there whenever you need it, and it combines all the options and benefits of audiographics conferencing and full motion video conferencing at the desktop or conference room through a single PC.

Fully Integrated Desktop Conferencing

VIS-A-VIS Video combines the shared space environment with video telephony on the same PC. Users can share graphic information and annotate in the shared screen at the same time as they can see and hear the person on the other end in full motion video.

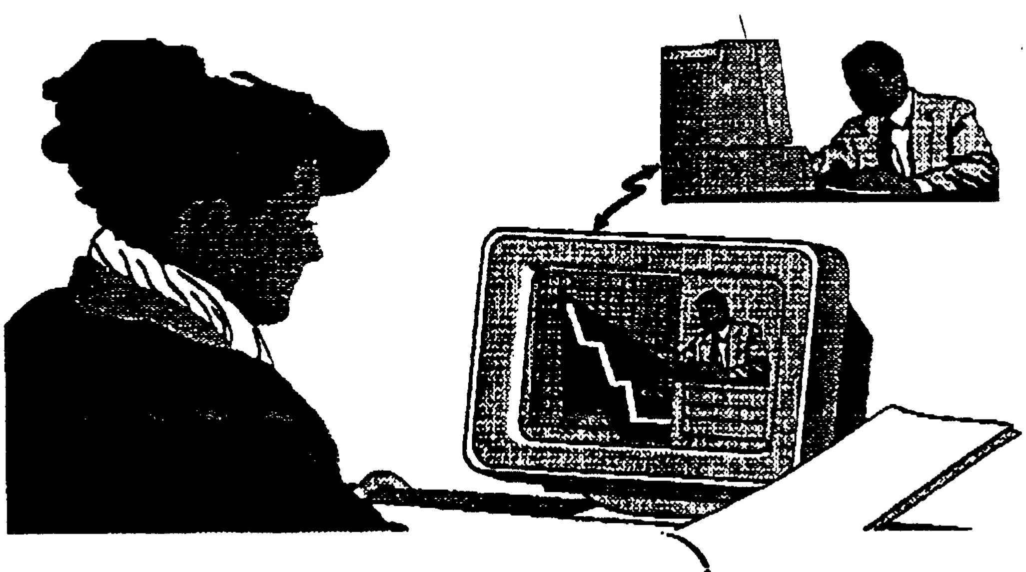
VIS-A-VIS Video does not restrict desktop conferencing to users with full motion video. Other VIS-A-VIS users can communicate with VIS-A-VIS Video users, but without the full motion interface.

Simple Controls For Ease of Use

The VIS-A-VIS Video screen is set-up so that the left 80% of the monitor contains the shared screen, with the user controls and the full motion video window on the right 20% of the screen. A set of Function selection controls allow any two of the Video, Folders, Pens, or I/O functions to be open at any one time in two open miniature windows.

The "Video" control overlays an image from an NTSC input source (Full Motion Video Codec, Camcorder, Camera or VCR) into the miniature full motion video window. The full motion video image can also be expanded into the shared space area of the screen or "zoomed" to a full screen image. A "Remote/Local" feature enables users to view the full motion video image being transmitted to them from a remote VIS-A-VIS Video site, or the image from their own site.

The "Folders" control provides four folders and a wastebasket from which slides can be stored, manipulated, and dragged to the shared space for display in a conference.



WORLDLINX

The "Pens" control gives users a selection of 3 colour pens, a highlighter, an eraser, a whiteout, or a keyboard for annotating in the shared space.

The "I/O" control displays a set of icons showing which Input/Output devices are available at your VIS-A-VIS site. These can include camera image capture, document scanner input or printer output.

Simple Full Motion Conversion

VIS-A-VIS Video is based on the standard VIS-A-VIS platform and I/O devices, and only requires the addition of a Videologic DVA card or a Truevision Bravado board to provide the NTSC overlay (full motion) function on the PC screen.

VIS-A-VIS Video works with widely available video codecs, and can be set-up to run in point-to-point connection through an RS232 interface on the codecs.

The VIS-A-VIS Video Benefits

- Low cost video conferencing on a PC
- Replaces expensive and time consuming distance face-to-face meetings
- Offers effective real-time interaction between people at different locations
- Facilitates fast, effective decision making

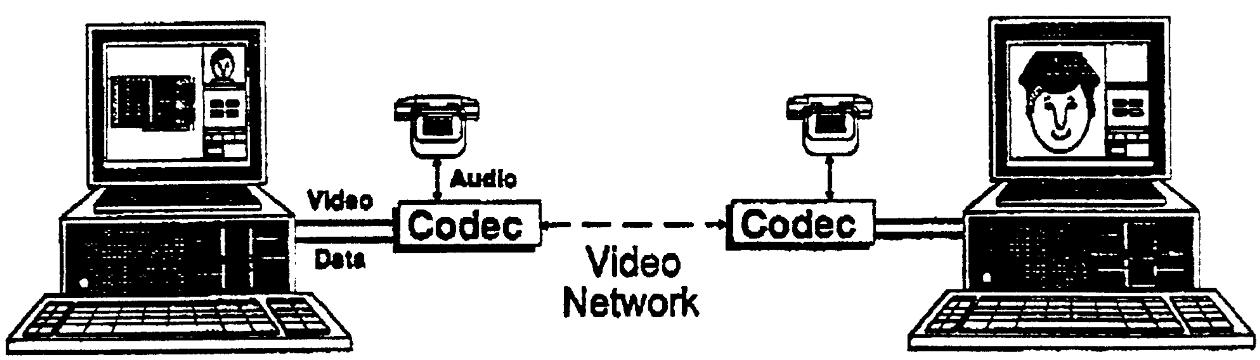
The VIS-A-VIS Video Advantage

- Full motion video information provided by a standard video codec connected to video network services such as 56Kbps, ISDN or fractional T1
- Offers full integration of video and data on your PC
- Provides access and control of various multimedia I/O devices such as document scanner, camera, laser printer and electronic whiteboard
- Offers standard data network connectivity support of VIS-A-VIS including data bridging

Numerous Applications

VIS-A-VIS Video offers numerous applications for business, government and education.

- Project control and management
- Manufacturing review
- Design/development review
- Telecommuting
- Distance education
- Full motion video conferencing
- Medical consultation and training



For more information, contact
VIS-A-VIS Sales & Marketing, 275 Matheson Blvd East, Mississauga, ON, Canada L4Z 1X8
Phone (416) 890-2773 FAX (416) 890-6789 Toll Free 1-800-263-9673

VIS-A-VIS

Configuration Guidelines

Basic Configuration

PC Requirements: IBM PS/2, IBM PC/AT or compatible, hard drive (min. 20 MB), 3MB RAM (4MB High Res.), Wacom tablet (or any compatible tablet and stylus) or Microsoft mouse or compatible, and MS-DOS V3.3 or later.
(Note: MS-DOS 5.0 recommended).

Monitors: VGA Monitor—16 colors, 640 x 480 pixel resolution, or
VGA Monitor—256 colors, 640 x 480 pixel resolution.

Multimedia Options

High Resolution: VIS-A-VIS supports 1024 x 1024 pixels, 16-bit color high resolution multisync monitor and Imagraph PM1010 card and cable. An 8-bit VGA card is also required for VGA support (Non-interlaced 32.5 to 64KHz). VIS-A-VIS has been tested with NEC, Hitachi and Taxan AGC monitors.
(Note: MCA High Resolution card not available.)

Scanner: VIS-A-VIS supports the CANON IX30 flat bed and HP ScanJet. Scanner interface cards are required.

Camera: VIS-A-VIS supports any RGB/NTSC/PAL input device like the JVC Camera, Camcorder, or the ELMO EV-308 Scanner. Your PC requires a full size 16-bit slot for a Targa-16 Card to drive these devices.

Printer: VIS-A-VIS supports the HP Laserjet II, IIP and Series III laser printers or any printer that supports HP's Laserjet Emulation. There are two ways to print documents from VIS-A-VIS—using a JLaser Card or through the parallel port on the PC. The JLaser Card prints an image in 15-20 seconds. Parallel printing takes up to 10 times longer. The JLaser Card requires a full size 16-bit slot in your PC.

Whiteboard: VIS-A-VIS supports the SMART Electronic Writing Board with an active area resolution of 3,500 by 2,600 pixels and an active display area of 120cm x 90cm. A projection system (such as an LCD panel and an overhead projector) projects the VIS-A-VIS image onto the board.

Video: VIS-A-VIS supports full motion video using the TrueVision Bravado board. Any codec providing NTSC output can be used with VIS-A-VIS. It has been tested with NEC, CLI and Picturetel codecs, as well as a PC codec on a card from VistaCom.
(Note: MCA High Resolution card not available.)

Note: VIS-A-VIS provides support for multimedia options at an additional cost. Each VIS-A-VIS workstation can be configured according to the user's individual needs.

WORLDLINX

Communication Options

Point-to-Point:

VIS-A-VIS can communicate point-to-point using synchronous, asynchronous, Netbios or TCP/IP communications.

Asynchronous:	Each PC requires a Hayes or compatible asynchronous modem (@ 2,400 to 19,200bps)
Synchronous:	Each PC requires an EICON Card with X.25 Network Access Software and a synchronous modem (@ 2,400bps to 64Kbps). Note: your PC requires an 8-bit slot for the EICON Card.
Netbios LAN:	Requires a Local Area Network card running a Netbios layer (acquire from your local LAN vendor).
TCP/IP LAN:	Uses PC/TCP by FTP Software (release 2.05 or later), or PathWay Access by Wollongong (release 2.0 or later).

Multipoint:

VIS-A-VIS can communicate multipoint over a LAN, packet-switched network, or the VIS-A-VIS Data Bridge.

The VIS-A-VIS Data Bridge is required for multipoint communications, or when you are communicating between different communication networks (e.g., synchronous to asynchronous communication).

The Data Bridge supports up to 8 simultaneous conferences with a total of 32 participants. A Data Bridge is not necessarily required for multipoint communication over a LAN or packet-switched networks, but performance can be impacted by the number of users, traffic and line speed, and a Data Bridge should be considered.

PC Requirements: IBM PC/AT or compatible, recommend 20MHz+, 2MB RAM, 40MB+ hard drive, monochrome monitor and card and MS-DOS V3.3 or later.

Appropriate synchronous, asynchronous and LAN interface cards are required.

Asynchronous:	Requires a Digiboard Eight Port (i.e., 8 users) Asynchronous Card. Your PC requires a full size 16-bit slot.
Synchronous:	Requires an EICON Four Port (4 users) Card with X.25 Network Access software. Your PC requires a full size 8-bit slot.
LAN:	Netbios and TCP/IP LAN requirements are the same as those shown under "Point-to-Point" section above.

For more information, contact
VIS-A-VIS Sales & Marketing, 275 Matheson Blvd East, Mississauga, ON, Canada L4Z 1X8
Phone (416) 890-2773 FAX (416) 890-6789 Toll Free 1-800-263-9673

Communique! System Requirements

Supported Platforms:	All Sun Microsystems® servers and workstations. All SPARC®-based servers and workstations. Hewlett-Packard Apollo 9000 Series 700 workstations. Contact InSoft for availability on other UNIX/RISC platforms.
Recommended Workstation Configuration:	32 Mb RAM. 5 to 10 Mb of temporary disk space. Full Color Display recommended.
Minimum Software Requirements:	SunOS 4.1.X with System V IPC facilities in system kernel or Solaris 2.X. OpenWindows® 3.0 HP/UX 9.0.1 or above and HP-VUE 3.0. 50 Mb swap space/60 or more preferred.

Communications/Network Requirements

Contact InSoft for specific configuration and bandwidth requirements.

Modem:	9600 baud minimum to run Communique! without video and audio.
Local Area Network:	Any TCP/IP or Ethernet based network.
Wide Area Network Services:	ISDN, SMDS, Frame Relay, Switched 56, FDDI, ATM 56 Kb to T1 and up.

Communique!TV Support

For Real Time, Full Motion Color Video Features –

Hewlett-Packard Series 700 Workstations:	HP VideoLive card using NTSC, PAL and SECAM signal formats.
Sun Workstations:	RasterOps SPARC TV II card using NTSC, PAL and SECAM signal formats. Sun VideoPix® SBus card using NTSC and PAL signal formats. Parallax Graphics® XVideo-24SV SBus card using NTSC, PAL and SECAM signal formats.

Let Communique! productively and profitably change the way you do business.
Call today for complete details and pricing – 717-730-9501.

InSoft
Revolutionizing The Way
You Do Business.

InSoft, Inc., Executive Park West One, Suite 307, 4718 Old Gettysburg Road, Mechanicsburg, PA 17055 Fax: 717-730-9504 email: info@insoft.com

VIEWPOINT SYSTEMS

Personal Viewpoint™ Product Brief

Overview

LAN/WAN Desktop Videoconferencing

Personal Viewpoint provides desktop videoconferencing capability to your personal computer. Operating under Microsoft Windows™ 3.1 in a 386 or 486 Personal Computer, the *Personal Viewpoint* is a single ISA expansion board. The board allows you to videoconference via your existing Ethernet or Token Ring LAN. Using the TCP/IP transport protocol, enables *Personal Viewpoint* to communicate via bridges and routers. *Personal Viewpoint* uses Viewpoint System's patented CPV™ (Compressed Packet Video) technology to provide variable bit rate, packet based communications designed to minimize LAN traffic. The video display is full color, up to 30 frames per second and completely sizable from an icon to full-screen. The processing power required for video compression and decompression is contained on the *Personal Viewpoint* board. *Personal Viewpoint* will display incoming video even while you are running other Windows applications. *Real-time* interactive video conferencing combined with other collaborative computing or groupware software products, provide productive electronic meeting capability.

Product Highlights

Personal Viewpoint performs *real-time* compression, decompression, display and packetization of live video for transport over local and wide area packet networks.

Capabilities include:

- *Real-time* video compression & decompression.
- Scalable color video display under Windows™ 3.1.
- Frame rate of up to 30 frames per second.
- TCP/IP transport via Ethernet or Token Ring.
- Variable Bit Rate Codec with Dynamic Burst Control.
- Less than 70ms of latency.

Background Display Capability

Personal Viewpoint will display incoming video even while it is not the active application. This allows you to work with a word processing document or a spreadsheet while you continue your videoconference. Any Window's application including Client/Server applications can be accessed during the videoconference.

Visual Caller ID

Upon receipt of a conference request, *Personal Viewpoint* user interface software will alert you with a dialog box indicating the name of the caller. You have the option of accepting or rejecting the call. In addition, you may choose to preview the calling party to visually confirm identity. After preview, you again have the option of accepting or rejecting the conference request.

Out of Band Audio

Due to the extremely low latency (less than 70ms) in the *Personal Viewpoint* video compression, audio can be transmitted out of band, while maintaining synchronization with the video. This allows the audio to be transmitted over existing telephone equipment.

Key Chat

Occasionally you may accept a videoconference request while you are already engaged in a telephone conversation. The Key Chat feature allows you to communicate with the video caller via the keyboard.

Video Mute

Video Mute functions much the same as audio mute on your telephone. When Video Mute is selected your video transmission stops. When Video Mute is deselected video transmission resumes.

Server Mode

Each *Personal Viewpoint* defaults into server mode upon Windows startup. There are several options with regard to incoming conference completion. The system can be configured to require you to confirm all conference requests. Listed known users or selected users from this list can be allowed automatic call completion. The system can also be configured for automatic completion of all conference requests. This feature is especially useful when you will be away from your desk for an extended period of time. Audio callers routed to your Voicemail can visually ascertain whether you are on the phone or away from your desk.

WinSock Compliant

Personal Viewpoint requires a PC TCP/IP stack. The system operates with any PC TCP/IP stack which supports the Windows Sockets API.

VIEWPOINT SYSTEMS

Personal Viewpoint™ Product Brief

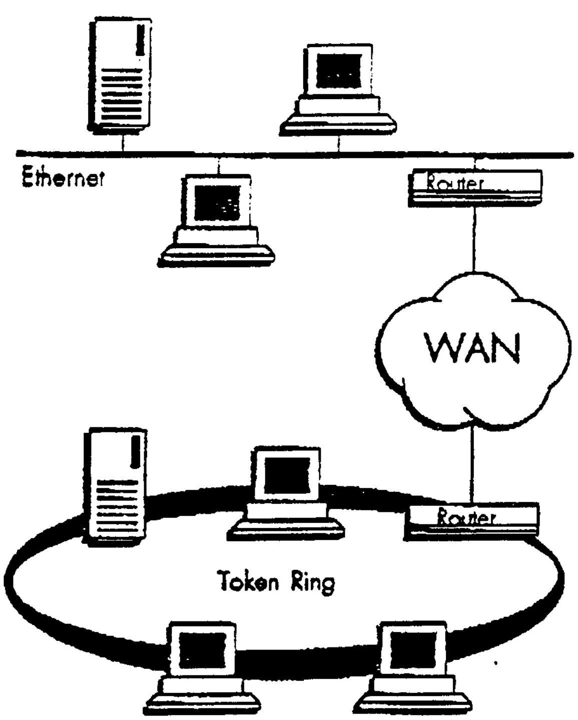
Dynamic Burst Control

Personal Viewpoint uses a variable bit rate codec. The maximum burst rate is configurable from 64 Kbps to 768 Kbps. If the network becomes congested, this can be dynamically adjusted, with no interruption to ongoing videoconferences. The maximum burst rate at each end of a videoconference can be configured differently.

Color Micro- Camera

The *Personal Viewpoint* Color Micro- Camera is a low lux, high resolution, color video camera, in a very small form factor. The camera is designed to be placed on top of your VGA monitor and manually pivots to adjust for monitor height. The camera can be deactivated for privacy. The video output of the camera is NTSC composite video. A user supplied NTSC composite camera may be used in place of the Color Micro- Camera.

LAN/WAN Desktop Videoconferencing



Specifications

Form factor:	13.4" x 4.2" (340mm x 107mm)
System requirements:	Intel386™DX 25MHz or better
Board interface connections:	
RGB video output:	DB-15 for VGA or multi-sync monitor
VGA feature connector:	VESA Standard
Video input:	RCA female connector NTSC composite
Video Resolution:	256H x 200V
Power consumption:	Nominal +5V .7A +12V .175A −12V .05A
Equipment Approvals:	FCC Class B

Personal Viewpoint is an ISA PC expansion board that provides *real-time* motion video compression and decompression for desktop video conferencing via packet networks. The compressed packet video can be transmitted over Ethernet, Token Ring, Frame Relay or FDDI.

Personal Viewpoint operates under Microsoft® Windows™ 3.1 with TCP/IP as the transport protocol. The end-user price for the *Personal Viewpoint* with color camera is $1,995.

2247 Wisconsin Suite 110 Dallas, Texas 75229 214.243.0634. Fax 214.243.0635

CONTACT : Richard Penn
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Suite 110
Dallas, Texas 75229
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ViewPoint Systems Introduces *Personal Viewpoint*™
Desktop Video Conferencing for Corporate LAN/WAN Networks

Editor's Summary

ViewPoint Systems, Inc., of Dallas. Texas has announced a new *desktop video conferencing* system called **Personal Viewpoint ™** that operates over existing PC Local Area and Wide Area Networks. The current release operates under Microsoft®Windows™3.1. A version that is compatible with Windows NT™ was introduced with Microsoft at Windows World '93 -- COMDEX/Spring '93 in Atlanta, May 24-27, 1993.

Dallas, Texas, Monday June 7, 1993 -- The new desktop video conferencing system for PCs, Personal Viewpoint, has a unit *selling price of $1995* including a color video camera. It *operates over existing Local Area and Wide Area Networks* (LAN & WAN) such as Ethernet™, Token-Ring™, FDDI, Frame Relay, ATM and other networks. **Personal Viewpoint** is a *plug-in option* for 386, 486, or P5-based IBM PC Compatible Computers with Microsoft Windows 3.1. Production delivery is scheduled to begin in July, 1993.

Personal Viewpoint uses existing, enterprise-wide, Local Area and Wide Area Networks, without any modifications required, to provide today's most cost effective method for conducting real-time, full-motion video conferencing. There are *NO* telephone company installation charges for wide bandwidth telecom lines, *NO* ongoing monthly connection charges; and *NO* time-based usage charges from a telecommunications carrier.

Glenn Norem, CEO of ViewPoint Systems, states "Personal Viewpoint delivers the productivity of today's video *tele*-conferencing *room* systems to the professional desktop. This new product empowers the individual with the video conferencing technology once reserved to only the senior executives of the largest corporations."

Operation and control of **Personal Viewpoint** is directed by the user from within the Microsoft Windows environment. Users can control the size and location of the full motion video window along with other application information displayed by the Windows 3.1 Graphical User Interface (GUI). The motion video is displayed at 30 video frames per second, and can be varied in size by the user from a full screen display to as small as a postage stamp-size icon. The user can choose to display the image being received or the image being transmitted at any time during the conference. Incoming video is displayed even while running other collaborative Windows applications.

--more--

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"Personal Viewpoint demonstrates that compressing full-motion video in real-time and delivering it via the established corporate-wide networks is a reality today! " said Alfred Riccomi, member of the Moving Pictures Experts Group as one of the U.S. Delegates to the ISO representing the X3L3 standards committee on MPEG.

Personal Viewpoint includes a plug-in ISA bus PC board, video conferencing software, a color camera small enough to be mounted on top of a VGA monitor, cables and connectors, and TCP/IP software for $1995. It can be installed into any 386, 486, or P5- based PC compatible system with a VGA display, which is connected to a Local Area or Wide Area Network (LAN &WAN). The add-in board accepts video input from any standard NTSC video source such as a camera, camcorder, cable TV, VCR or Laser Disc player. **Personal Viewpoint** is available without the camera for $1595.

ViewPoint Systems' policy is to comply with established network standards. The **Personal Viewpoint** desktop video conferencing system conforms with the TCP/IP industry standard. **Personal Viewpoint** operates with network interface boards that conform to IEEE STD-802.3 Ethernet and IEEE STD-802.5 Token-Ring standards. ViewPoint Systems is one of the leaders in an effort to establish an industry-wide association to define protocols and standards for video conferencing over packet-switched networks (eg. LANs & WANs). Network gateways are planned to interconnect **Personal Viewpoint** video conferences conducted over packet-switched networks to video conferencing terminals operating over circuit-switched Telecom Networks using the CCITT International Standard H.261 (also known as px64).

The **Personal Viewpoint** system uses an innovative packet video compression technology. It operates over existing, packet-switched LAN and WAN networks at data rates selectable by the users and *Network Managers* ranging from 64 kilobits per second up to 768 kilobits per second. The video frame rate is selectable by the user for a *constant 30 video frames per second* or a variable frame rate mode of operation.

Personal Viewpoint extends the video communications product offerings of ViewPoint Systems, Inc., which includes group conferencing systems that operate over both circuit-switched Telecom networks and existing packet-switched Local Area and Wide Area Networks.

VideoView

Personal Technology Re-

Special Report

May 17, 1993

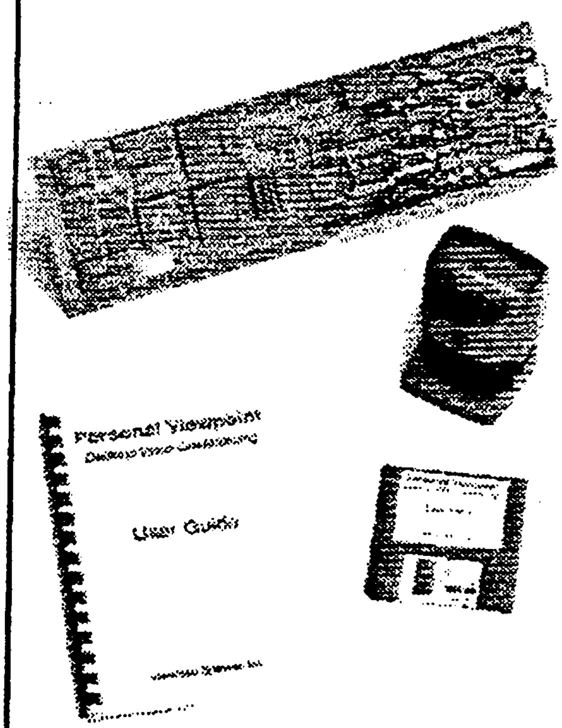
ViewPoint Systems Focuses on LAN Environment, Announces Personal Viewpoint (TM) Desktop Videoconferencing System

The Old and the New.... the older model of videoconferencing via the switched telecommunications network - and the new - real-time video via the LAN. While some players in the videoconferencing industry have been slow to focus on the significance of packet video over the LAN, it is clear that a next generation of videoconferencing equipment providers is emerging to address these issues associated with LAN to telecommunication network connectivity.

The newly formed ViewPoint Systems, originally as Bolter Communications, has been working with PC-based videoconferencing solutions over packet networks with companies including Sprint, MCI, Northern Telecom and WilTel. The company is not new to the world of packet video and had developed the basic compression technology to support videoconferencing capabilities over frame relay. ViewPoint Systems has recently introduced its premier desktop board-level product, Personal Viewpoint.

Personal Viewpoint is a single ISA 386/486 PC expansion board which enables real-time motion video compression and decompression via packet switched networks using the transport protocol TCP/IP supporting communication over LANs, bridges and routers. Compressed packet video can be transported across the Internet, Ethernet, Token-Ring, Frame Relay and FDDI.

The total price including the color camera, software and manual is $1,995. Windows (TM) 3.1 and a PC TCP/IP stack are required. Additionally, ViewPoint Systems will announce a version of the product at Windows World/ COMDEX-Spring '93 this month that is compatible with Windows NT, a Microsoft/



Published by:
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Waltham MA 02154
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FAX (617) 893-6334

Intel initiative to link the PC environment with the telecommunications environment.

Product Overview
Personal Viewpoint uses the company's Compressed Packet Video (CPV) technology supporting variable bit rate coding techniques to achieve temporal and spatial scalable video compression. Packet networks operate asymmetrically, e.g. data may be sent and received at different and dynamically changing data rates dependent on network traffic.

CPV looks at each frame of video and sends only those pixels which have changed significantly between frames and which are necessary to maintain the integrity of the image. According to ViewPoint Systems, CPV is responsive to the amount of bandwidth available in determining which pixels will be transmitted e.g. a high bandwidth pipe enables a greater number of pixels to be sent and therefore fewer artifacts.

The company states that the resolution of transmitted video images is 256H x 200V at 30fps. Users can select either the constant 30fps video frame or variable frame rate operation.

According to ViewPoint Systems, the maximum burst rate is configurable from 64kbps to 768kbps and bandwidth parameters can be dynamically adjusted during an ongoing videoconference. The packet video implementation is reported to have extremely low latency at less than 70ms and therefore will support out-of-band (telephone handset) audio.

Importantly, Personal Viewpoint enables the user to work on an active Windows application while participating in an ongoing videoconferenence. Any Windows application, including client/server applications may be accessed during a videoconference.

Other features include: Visual Caller ID, "Key Chat", Video Mute and Server Mode.

Market Strategy
ViewPoint Systems plans to market Personal Viewpoint via established "LAN 100" distribution channels who have significant expertise in the sales, installation and support of datacom type products within the LAN/WAN enterprise network environment of major Fortune 2000 companies. Additionally, the company intends to encourage and support teaming agreements with established software vendors so that it will be positioned to deliver applications and tools demanded in the videoconferencing marketplace for vertical applications.

Support of the Compressed Video Interoperability Protocol (CVI)(1)
Principals of ViewPoint Systems have publicized the question which is on every ones minds these days: Now that real-time and stored video transmission capabilities have come to the desktop via computer and communications networks, how do we create working environments which enable different platforms and different vendors terminal devices to "talk" with one another? How can we "grow" the visual communications market if the industry itself creates islands of visual communications users?

This is a big question because in the world of video transmission, multiple coding schemes for compressed digital video is determined by specific vendor hardware platforms and needs. If governed by this paradigm, distinct user groups have access to specific systems only, while to truly grow the market for this technology, video communications must approximate a percentage of the availability and interoperability as is characteristic of the regular telephone.

Users who have invested in a system which supports visual telecommunications via PSTN at Px64 should not have to abandon use of their system to videoconference with users of a packet video systems as that offered by ViewPoint. Therefore, the step towards the acceptance of a standard protocol supporting the interoperability among as many visual communications systems is as is possible and appropriate, is critical. According to ViewPoint Systems, which is taking a lead in CVI activities, the capability of the CVI Protocol will be demonstrated before the end of the summer 1993.

(1) Riccomi, Alfred and Norem, Glenn, Interoperability for Incomparible Video Codecs: The "CVI" Protocol, presented at MPEG Committee Meeting, March 1993.

INVISION™ Video Teleconferencing for Windows

InVision™ Video Teleconferencing for Windows is the result of an evolutionary approach developed by InterVision Systems Corporation to deliver an affordable audio and video teleconferencing solution to your very own networked, Microsoft Windows 3.1-based PC.

Unlike other video teleconferencing solutions that require expensive and dedicated, single-function, non LAN/WAN-based workstations, as well as costly digital telephone lines or satellite links, InVision Video Teleconferencing for Windows uses the existing computer network infrastructure to exchange real-time audio and video information. Your existing LAN/WAN is no longer limited to simply transmitting and receiving data/text files and character-based electronic mail messages. InVision gives you and your colleagues a long-awaited and very valuable communications tool that is designed to directly increase your daily effectiveness and productivity. InVision is designed for you and everyone else on your network!

Increase Corporate Efficiency

With InVision, corporate network users now have a powerful tool that extends the power of their installed computer networks beyond file transfers and exchanging text messages. Now users can have face-to-face meetings, across the corporate campus or across the company, without ever leaving their office. InVision users will realize significant savings in travel time and increased efficiency.

The most effective way to express your ideas and measure their immediate and spontaneous impact is through visual contact, and such interaction can be yours with InVision Video Teleconferencing for Windows.

This is what you can expect from InVision Video Teleconferencing for Windows:

- Increased productivity and efficiency
- Reduced out-of-office time
- Faster and better decision-making cycles
- More productive meetings
- Reduced travel time

Compatibility is the Key

InVision has evolved from the new generation of **Codec (compression-decompression)** technology for the personal computer. The InVision video teleconferencing solution consists of an Intel i750™ based Codec board, a color video camera, and InVision software. With this combination of hardware and software, any 80386/486-based PC can be upgraded to support real-time videoteleconferencing over any computer network platform: Ethernet, Token Ring, FDDI, or ISDN.

The InVision communications protocol is built on the TCP/IP model, but since the product uses a Virtual Sockets Library™ API, it supports any vendor's TCP/IP stack. Support for additional communications protocols is expected in future releases. And since InVision has been developed using a Virtual Sockets Library API, it is also 100-percent compatible with the Windows Sockets API standard.

INTERVISION
SYSTEMS CORPORATION

Overall Features

InVision Video Teleconferencing for Windows is a complete solution which:

- Allows full-motion, real-time color video and audio teleconferencing.
- Utilizes existing Local or Wide Area Network and ISDN infrastructures.
- Uses the international standard TCP/IP as the transport protocol.
- Uses Intel DVI™ and i750-based technology, a global de facto standard for desktop video.
- Provides phone/address book support.
- Operates independently of other Windows-based applications.
- Uses Microsoft's Video for Windows compatible hardware.
- LAN independent: Ethernet. Token Ring, FDDI and ISDN.
- Supports both NTSC and PAL video formats.

Video and Connectivity Features

InVision Video Teleconferencing for Windows offers a full range of video and connectivity functionality:

- Bidirectional, full-duplex audio and video transmission.
- Bandwidth configuration routines to support data transmission rates of 128Kbps, 256Kbps and 384Kbps.
- Adjustable Volume. Brightness, Contrast, Tint and Color saturation.
- Uses Host Table as the Address Book functions to maintain names.
- Video Capture capability to capture single frames or sequences of frames to disk.
- Capture images of remote documents immediately.
- Move and resize the video window to fit your desired workspace and screen size.
- Control your video with on-screen "mute" button.
- Review slide presentations on-screen.
- Transfer files, access remote systems, log-on to other systems — all during the live video teleconference session.
- Operates alongside other MS Windows applications.
- Windows-based Installation.
- Context-sensistive online help.

System Requirements

- Personal Computer with Intel 386DX/33 Mhz or higher processor running Microsoft Windows operating system 3.1
- 3.5" high-density (1.44MB) disk drive
- 8 MBytes of RAM
- 3 MBytes of available disk space
- any VGA graphics card with a feature connector (or SVGA card running in VGA mode)
- Microsoft Mouse or compatible pointing device
- One 16-bit expansion slot

Network Support

InVision Video Teleconferencing for Windows is fully compatible with the following network adapters:

- 3Com
- IBM
- Intel
- Novell
- SMC
- Any adapter with ODI or NDIS Drivers

InVision operates over existing TCP/IP based networks, including:

- 3Com 3+ Open TCP
- Beame & Whiteside BW-TCP and BW-NFS
- Distinct Software TCP/IP
- D-Link TCP/IP for DOS
- Frontier Technologies Super-TCP/IP
- FTP Software PC/TCP
- HP ARPA Services for DOS
- Locus TCP/IP for DOS
- Microsoft LAN Manager TCP/IP
- NetManage Chameleon & ChameleonNFS
- Novell LAN WorkPlace for DOS
- Spry Air for Windows
- Sun PC/NFS
- Ungermann-Bass Net/One
- The Wollongong Group PathWay Access and Win TCP for DOS
- Any generic Windows Sockets-based TCP/IP

Product Includes

InVision Video Teleconferencing for Windows includes:

- InVision Software - Windows DLL (Dynamic Link Library)
- Intel i750-based Codec board
- Color Video Camera and cables
- External Microphone and Speaker (optional)
- Documentation

Support

InVision Video Teleconferencing for Windows comes with 90 days of telephone technical support. Annual software maintenance support is available as an option.

Call for free interactive multimedia brochure for InVision.

InVision is fully compatible with Microsoft® Windows™, Intel®, DVI® multimedia, Indeo™ Video technology, and i750™.

InterVision Systems Corporation
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Telephone: (703) 560-1446 • Fax: (703) 560-4364 • Internet: wk05020@worldlink.com

INTERVISION
SYSTEMS CORPORATION

APPENDIX 3.2

PACKETIZATION OF H.221 FRAMES FOR LANs

ITU Telecommunication Standardization Sector | Document AVC-587
Study Group 15 | October 1993
Experts Group for Video Coding and Systems in ATM and Other Network Environments

SOURCE: Stephen Hall, Monash University (Australian UVC Consortium)

TITLE: Packetisation of H.221 frames for LANs

PURPOSE: Discussion

Abstract

A means of adapting H.320 terminals for use on packet switched LANs, in which the H.221 frame structure is retained, is discussed. The appeal of this approach is that it allows maximum use to be made of existing H.320-related recommendations and equipment. However, the effect of packet loss is identified as an important unresolved issue, requiring further study.

1. Introduction

The adaptation of H.320 audiovisual terminals to packet switched LANs has been considered in AVC-512 and AVC-513. In AVC-512, it was concluded that the replacement of the H.221 multiplex and framing structure with an alternative, H.22z, would provide the greatest benefits in terms of performance and flexibility. Nevertheless, it was recognised that the design of H.22z was a substantial task, and that this approach might preclude the use of existing H.320 hardware in H.32z terminals. The alternative H.221-based approach is therefore discussed further in the present document, in order to provide greater insight into the relative merits of the two approaches.

2. Network configuration

An example network configuration is shown in Fig. 1. In this scenario, an H.32z terminal connected to a LAN is able to communicate with another H.32z terminal on the same LAN, as well as with a remote H.320 terminal connected to an ISDN network. Interworking with an H.32x terminal connected to a B-ISDN network is also possible, assuming that the H.32x terminal has an H.320 compatibility mode.

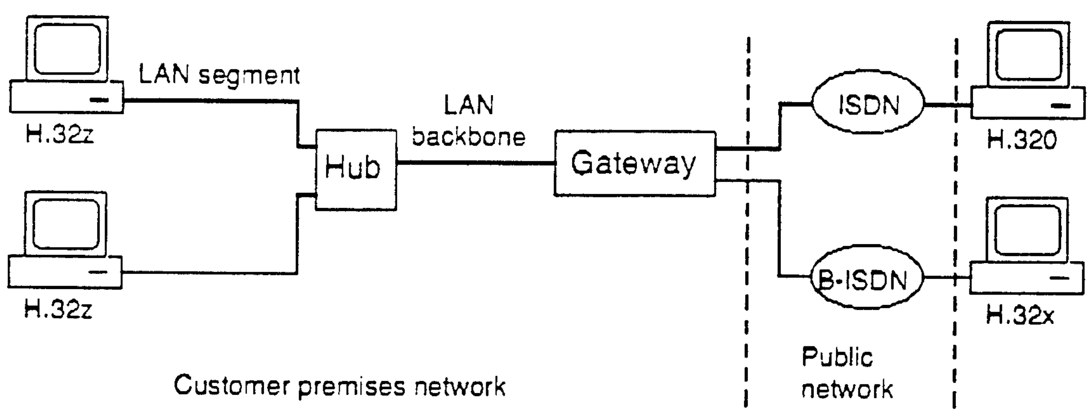

Fig. 1 An example network configuration

The function of the gateway is to terminate the packet protocols used on the LAN. However, it does not need to perform any special operations on the H.221 bit stream itself (such as remultiplexing), so its processing load is relatively low. It is therefore likely that existing LAN data gateways could be adapted for this application by means of a software upgrade.

2. Protocol stack

A possible protocol stack in an H.32z terminal is shown in Fig. 2. At the lowest layer, the LAN protocol controls access to the physical medium, and provides basic functions such as framing and bit error detection. Above this, a "packet adaptation layer" implements the additional functions required to handle real-time traffic, such as timing recovery. It thus appears to a device above the adaptation layer (in particular an H.320 terminal), that it is talking directly to an ISDN terminal adapter.

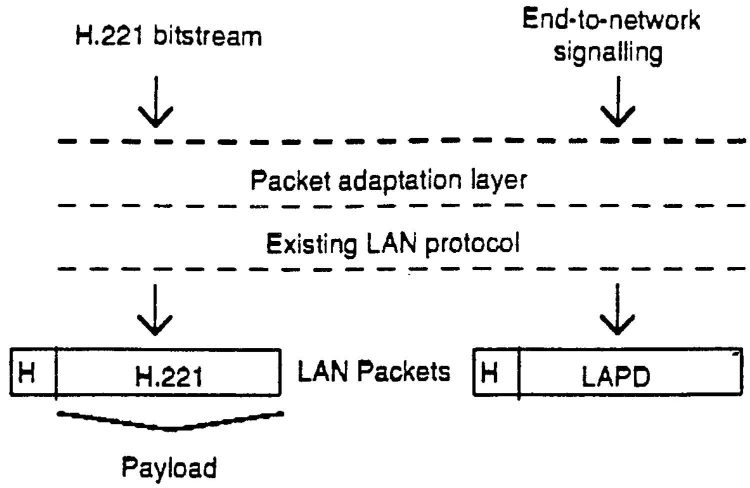

Fig. 2 Possible protocol stack in an H.32z terminal

In addition to the H.221 bitstream, it is necessary to provide a path for the end-to-network signalling (in particular, call control signalling) from the H.32z terminal, across the LAN, to the ISDN network. This can be done by encapsulating ISDN D-channel messages within LAN packets.

3. Packet adaptation layer design issues

Data Alignment

An H.320 terminal transmits and receives streams of data at a constant bit rate. The simplest way of carrying this information across a packet switched LAN is to segment a bit stream into blocks of a suitable size (say a few hundred bytes), and then to put these blocks into packets. It may at first appear that the packetisation process should be aligned to the H.221 framing structure, so that a packet always contains an integral number of frames. This technique can improve loss resilience when applied to the packetisation of individual media streams (eg. aligning packets to video GOBs, as described in AVC-513). However, in the configuration described here, the media streams have already been multiplexed into H.221 frames in such a way that there is no alignment to media units. Furthermore, the fact that the H.221 frames are of a constant length makes it relatively straightforward for the packet adaptation layer to maintain the H.221 frame structure in the event of packet loss, provided it knows how much data has been lost. For these reasons, packet/frame alignment is not necessary, which gives greater freedom in the choice of the packet size.

Packet size

The packet size on popular LANs (eg. Ethernet, Token Ring, FDDI) can be chosen by an application from a fairly wide range, typically 64 - 1500 bytes, with a packet header of 20 - 30 bytes. The need to minimise the packetisation delay at low bit rates leads to the requirement for fairly short packets, in the region of a few hundred bytes. For example, when the H.221 bit rate is 128 kbit/s, a packet payload of 320 bytes takes 20 ms to accumulate. The packetisation delay is incurred at each point where data enters the LAN, ie. once in the transmit direction and once in the receive direction in Fig. 1.

In general, successive LAN packets may have different sizes. However, in this application, a fixed packet size is appropriate, as this corresponds to a constant packetisation delay, and makes it easier to decide how much dummy data should be generated to replace a lost packet.

Packet loss

LAN protocols automatically perform error checking on both header and payload bits, and corrupted packets are discarded. However, LANs typically have very low bit error rates, in the region of 10^{-9}, so that packet losses due to bit errors are correspondingly rare. Packet losses due to queue overflows are more likely, but can be minimised by good network design and management. In any event, the loss of a packet containing H.221 data is equivalent to a lengthy error burst on an ISDN channel (assuming that the lost packet is

replaced with dummy data). This may have a severe subjective effect on the communication, particularly if it causes a loss of H.221 frame or multiframe alignment in the H.320 terminal. It is to be expected that the packet size would have a strong correlation with the effect of packet loss, but no quantitative results on this issue are available at present.

Packet loss can be detected in the LAN environment by means of sequence numbers in the packet headers. It may therefore appear that the effect of packet loss on the video signal could be mitigated by initiating a picture refresh through a "Fast Update Request". However, this BAS code can only be sent in the H.221 service channel, which is not available to the packet adaptation layer.

In contrast, the loss of a LAN packet containing end-to-network signalling will be taken care of by the D-channel protocol, which will automatically perform retransmission.

Timing recovery

As the transport time on packet switched LANs is variable, it is necessary to buffer the received packets in the adaptation protocol, in order to recreate a constant bit rate stream. The extra delay incurred in this process can be kept small by minimising the packet jitter through appropriate design and control of the LAN.

The fact that the ISDN network clock is not directly available to the H.32z terminal must also be taken into account. This means that the H.32z terminal may produce data faster than it can be transmitted on the ISDN link, leading to data loss. A solution is to generate a suitable clock signal within the packet adaptation layer, and to lock this to the ISDN network clock, by passing flow control information across the LAN in the packet headers.

4. Conclusion

One approach to creating an H.32z terminal, in which the H.221 bitstream and signalling messages from a conventional H.320 terminal are packetised, has been discussed. The advantages are that maximum use is made of existing H.320-related recommendations in the H.32z design, and that existing H.320 equipment may be converted to H.32z by means of an "add-on" adapter. However, the subjective impact of packet loss is unknown, and further study of this issue is therefore required before a conclusion can be reached on the feasibility of this approach.

References

AVC-512 (Australian UVC Consortium), "Design of H.32z terminals for LANs", July 1993.

AVC-513 (Australian UVC Consortium), "H.22z protocol for LANs", July 1993.

APPENDIX 3.3

H.22Z PROTOCOL FOR LANs

Telecommunication Standardization Sector
Study Group 15
Experts Group for ATM Video Coding
(Rapporteur's Group on Part of Q.2/15)

Document AVC-513
July 1993

SOURCE: Stephen Hall (Australian UVC Consortium)

TITLE: H.22z protocol for LANs

PURPOSE: Proposal

Abstract

This document describes an H.22z protocol for transporting real-time audiovisual/multimedia information over packet switched local area computer networks. The required functions are identified, and a corresponding packet structure is proposed.

1. Introduction

Document AVC-512 discusses the basic design of an H.32z terminal, which is intended for connection to packet switched local area networks (LANs). The present document describes an associated H.22z protocol, which provides the terminal with a means of conveying multimedia information across the LAN. The H.22z protocol assumes that the LAN is able to transport packets between two or more network nodes within suitable delay bounds, but with no guarantee of delivery (ie. using an "unacknowledged datagram" service). Such a service can be offered on suitably configured Ethernet, Token Ring, FDDI and ATM LANs, which incorporate basic resource allocation mechanisms in network switches and routers. Examples of unacknowledged datagram protocols include IEEE 802.2 LLC Type 1, and Internet IP. The H.22z protocol adds functions to enhance the basic datagram service as appropriate. For example, it implements an automatic retransmission scheme to correct errors in end-to-network signalling streams. A layered protocol model incorporating H.22z is shown in Fig. 1.

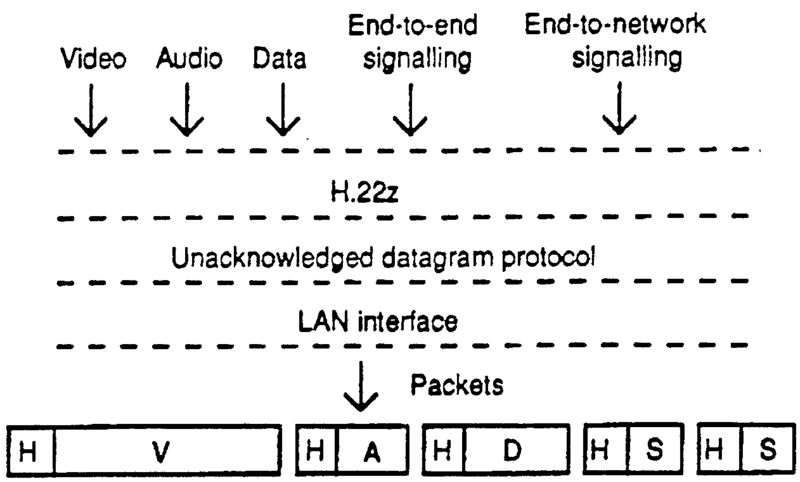

Fig. 1 Layered protocol model incorporating H.22z

2. Functions of H.22z

- Packetisation: The H.22z protocol must segment the various data streams into units which are suitably sized for transport on the LAN. The length of each LAN packet can be matched to the bit rate of the associated data stream, in order to minimise the packetisation delay. Where a natural data unit, or "message", is present in the stream, such as a video GOB or slice, then it may be advantageous to align the packets with these data units. For example, the effects of packet loss on video quality can be reduced by aligning GOBs/slices to packets. However, since a long message may span more than one packet, an indication of the last packet in the message is required, in order to minimise the reassembly delay. This can be provided by a "segment type" field in the H.22z packet header, which indicates that

the packet contains the beginning of a message (BOM), the continuation of a message (COM), the end of a message (EOM), or a single segment message (SSM). In addition, it is possible that messages will not be aligned to *byte* boundaries in the packet payload. An example is where the payload is created by segmenting a continuous bit stream from a video codec. The provision of pointers in the H.22z packet header to the first and last bits in the payload avoids the need for time-consuming bit-shifting operations prior to packetisation.

- Multiplexing: As described in document AVC-512, the H.22z protocol is required to provide a multiplexing function for the various data streams from a single terminal. Furthermore, a packet multiplex approach, where each packet contains only one type of data, is shown in AVC-512 to be preferable to one based on the encapsulation of H.221 frames. This leads to the requirement for a "stream identification" field in the H.22z packet header. The field should be sufficiently large to allow the identification of streams from multiple terminals, in order to accommodate future multipoint configurations.

- Error handling: Packets may be lost on LANs due to bit errors or queue overflows. Bit error rates on LANs are very low, typically 10^{-9}, and queue overflows can be avoided through proper network design and management. Nevertheless, the losses which do occur must be detected by the H.22z protocol, and this can be done using a sequence number in the H.22z packet header. In addition, appropriate action must be taken once an error has been detected. In the case of lost audio or video information, the terminal must be informed of the occurrence and location of the loss, in order to allow it to employ its own error concealment or recovery techniques (eg. a "fast update request"). In contrast, lost end-to-network signalling information can simply be retransmitted. However, special measures must be taken to avoid the loss of end-to-end signalling and C&I information (eg. BAS), since this may be delay-sensitive. For example, packets containing such information can be duplicated prior to transmission.

- Timing recovery: Most LANs operate asynchronously, implying that there is no fixed timing relationship between successively delivered packets. Furthermore, there is often no master network clock which can be used as a timing reference by the terminals. The H.22z protocol must therefore provide a means of recovering the timing between packets, and synchronising the terminals. This can be done using a combination of buffering and sliding window flow control, leading to the requirement for a numbered acknowledgement field in the H.22z packet header.

3. Packet structure

A structure for H.22z packets, designed according to the above requirements, is shown in Fig. 2. Note that the H.22z packet header is only 3 bytes long, while the payload may be several hundred bytes in length, so that the packetisation overhead is very low.

SID	SEQ	ACK	ST	PF	PL	PAYLOAD

SID = Stream identifier (8 bits)

SEQ = Segment sequence no. (4 bits)

ACK = Acknowledgement no. (4 bits)

ST = Segment type = BOM, COM, EOM or SSM (2 bits)

PF = Pointer to first bit in payload (3 bits)

PL = Pointer to last bit in payload (3 bits)

Fig. 2 H.22z packet structure

4. Conclusion

A simple protocol for transporting multimedia information from H.32z terminals across packet switched LANs has been proposed. Further work is required to specify its operation in detail.

APPENDIX 3.4

DESIGN OF H.32Z TERMINALS FOR LANs

Telecommunication Standardization Sector
Study Group 15
Experts Group for ATM Video Coding
(Rapporteur's Group on Part of Q.2/15)

Document AVC-512
July 1993

SOURCE: Stephen Hall (Australian UVC Consortium)

TITLE: Design of H.32z terminals for LANs

PURPOSE: Proposal

Abstract

The demand for access to multimedia services from LAN-attached desktop computers is increasing, and a standards-based solution will ensure maximum connectivity is achieved, through interworking with existing and future TSS SG15 terminals. Two options for the design of H.32z terminals are proposed, and it is concluded that the scheme in which the various data streams are multiplexed into separate LAN packets offers the greatest benefits.

1. Introduction

There is a growing demand from users for access to audiovisual/multimedia services, using desktop computers which are attached to packet switched local area networks (LANs). While proprietary solutions suitable for local or closed-user-group communications are emerging, wide connectivity can only be achieved through a standards-based approach. In particular, interworking with terminals covered by current and future TSS SG15 recommendations (eg. H.320 and H.32x) is required. With this in mind, and considering the many similarities between packet switched LANs and ATM networks, we believe that the Experts Group is in the best position to develop the appropriate recommendations (H.32z, H.22z, etc.).

Existing and future packet switched LANs are likely to incorporate a range of networking technologies, including Ethernet, Token Ring, FDDI and ATM. The common denominator in such LANs is the ability to transport variable-length packets asynchronously between network nodes (involving segmentation and reassembly in the case of ATM). By configuring the network appropriately, and using bandwidth allocation mechanisms in network routers and switches, the delay constraints imposed by real-time multimedia traffic can be satisfied.

An example network configuration is shown in Fig. 1, where two H.32z terminals are connected to a high-speed LAN backbone via dedicated Ethernet and Token Ring segments. The LAN hub allocates bandwidth for the real-time traffic and switches packets between the different LAN segments. A gateway between the LAN and the public network allows communication with remote H.320 and H.32x terminals, by converting between the H.22z protocol used on the LAN and the H.221 or H.22x protocols used in the public network. Note that the gateway is part of the customer premises network, but it has standard ISDN and B-ISDN network interfaces.

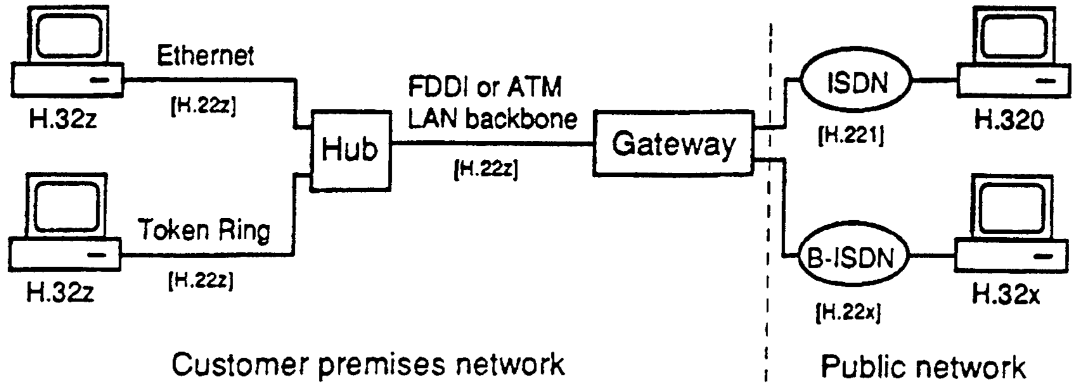

Fig.1 Example network configuration.

2. H.32z Terminal Design

We consider below two ways of creating an H.32z terminal, by converting an H.320 terminal for operation on a packet switched LAN. Essentially, these involve making a "cut" through the existing design in one of two places (marked A and B in Fig. 2), and implementing a new H.22z protocol for transporting the multimedia information across the LAN. This approach makes substantial use of modules covered by existing recommendations, and should therefore enable the realisation of H.32z terminals in the near future.

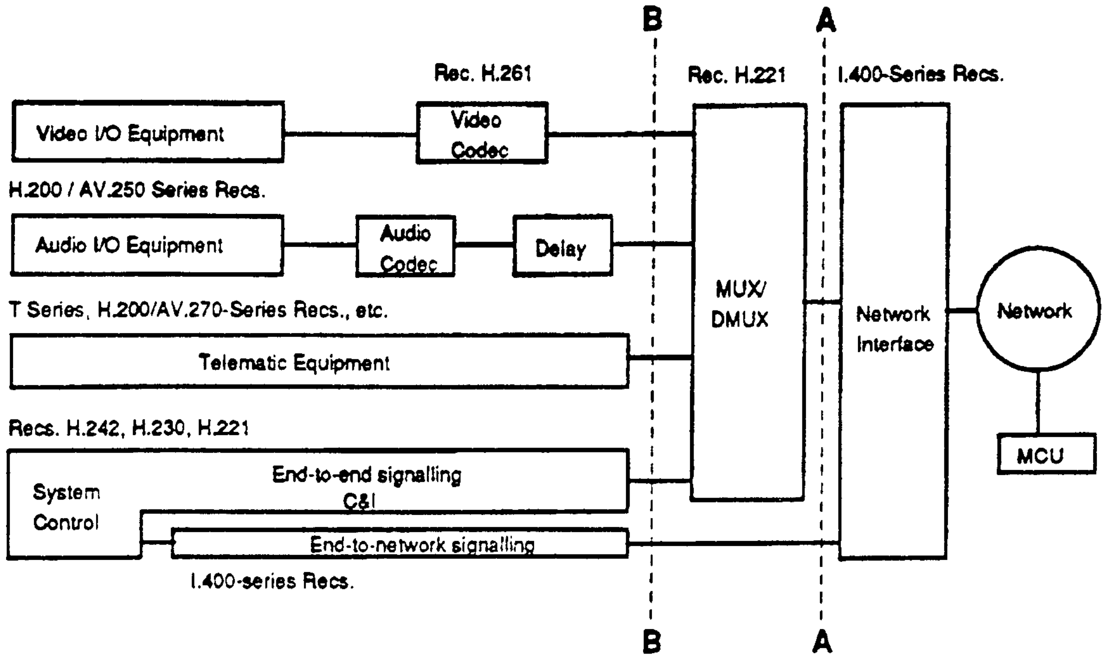

Fig. 2. H.320 terminal structure [from TSS Rec. H.320, Figure 1/H.320, 1990]

Scheme A:

In this approach, the H.221 framing function is retained in the H.32z terminal, and one or more H.221 frames are placed into packets for transmission across the LAN. The benefits of this approach are:

- There is minimal disruption to the existing H.320 terminal design.
- The temporal relationship between the various data streams established by the H.221 frame is implicitly maintained across the LAN.

Scheme B:

In this approach, the multiplexing, framing and network interface functions which are specific to ISDN (or B-ISDN) are omitted from the H.32z terminal, and placed in the gateway. Each data stream (video, audio, signalling, etc.), is then transmitted separately across the LAN, ie. using a different LAN packet for each type of data. Scheme B offers the following benefits:

- Different data streams can be provided with the appropriate quality of service on the LAN. For example, automatic retransmission can be provided for corrupted or lost end-to-network signalling information, whereas the extra delay incurred makes this inappropriate for audio and video.
- There is scope for implementing advanced service features by making use of the unique characteristics of the LAN environment. For example, multicast packet addresses can be used to send video and audio information simultaneously to a number of terminals on the LAN. Similarly, an audio stream can be separately routed to a terminal which does not have a video capability.

- Any increase due to packetisation in the end-to-end audio/video delay can be minimised, by varying the LAN packet length according to the occupancy of the video coder output buffer. In this case the buffer will be emptied in bursts, so that it will be necessary to perform the video rate control function using a "virtual" buffer (ie. an up/down counter), rather than the physical buffer.

- The H.32z terminal is more amenable to incorporation in desktop computers, since the H.221 bit-oriented multiplex is replaced by a byte-oriented approach. This facilitates access by application software to signalling and user data in the multimedia streams.

- Extension to higher bit rates is facilitated. For example, it may be desirable in the future to create an H.32z terminal which can handle a bidirectional video stream at CIF resolution for conversational services, and a unidirectional stream at a higher spatial resolution, for retrieval services. The flexible nature of packet-oriented multiplexing allows such enhancements to be added without redesigning the multiplex.

3. Conclusion

Two ways of designing an H.32z terminal for use on packet switched LANs have been described. On balance, the benefits offered by scheme B, in which the individual data streams are multiplexed into separate LAN packets, are considered to outweigh those offered by scheme A, which is based on the encapsulation of H.221 frames. An H.22z protocol designed according to scheme B is proposed in AVC-513. Further work is required to establish whether other portions of the H.320 terminal (eg. H.242, H.230) need to be modified for the LAN environment. In addition, the potential for harmonisation of H.32z with H.32x and associated recommendations should be considered.

APPENDIX 3.5

VIDEO AND AUDIO COMMUNICATION SYSTEMS FOR CSMA/CD LAN

ITU Telecommunication Standardization Sector
Study Group 15
Experts Group for Video Coding and System
in ATM and Other Network Environments

Document AVC-595

October 21, 1993

Source : NTT
Title : **Video and Audio Communication System for CSMA/CD LAN**
Purpose : **Information**

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1. Introduction

NTT has developed a realtime video and audio communications system for 10 Mbps CSMA/CD LAN. This system consists of an audio-video realtime communication adapter and ISDN visual telephone gateway. This adapter establishes connection by TCP protocol, and transmits audio and video data with 16-208 kbps by UDP. ITU-T Rec. H.261 and G.728 coding algorithms are used for coding/decoding video and audio respectively. For real-time audio-video communication over LANs, congestion control and video frame refresh controls are implemented. The ISDN visual telephone gateway realizes interconnection between the audio-video real-time communication adapter and ISDN visual telephone terminal based on ITU-T rec. H.320 terminal.

2. Service concepts

Fig.1 shows the audio-video tele-conference system. PCs, WSs and the audio-video real-time communication adapters are connected with 10 Mbps CSMA/CD LANs, LAN-to-LAN connections can be made via basic- or primary-rate ISDN circuits. ISDN visual telephone gateway is for interconnection between the adapter and ISDN visual-telephone based on ITU-T H-series Recommendations.

(1) Audio-Video real-time communication over LANs and WANs

The adapter can establish a connection with another adapter over LANs or WANs. Transmission rates of audio and video are 16 kbps, and 16-192 kbps variable, respectively. The maximum transmission rate of the pair is 208 kbps. When 5 pairs of adapters with 208 kbps transmission rate are communicated at the same time, it occupies about 2.1 Mbps of 10 Mbps CSMA/CD LAN capacity. If these terminals are set at 144 kbps (128 kbps for video and 16 kbps for audio), total bitrate of audio-video communication is about 1.4 Mbps.

(2) Interconnection between audio-video real-time communication adapter and ISDN visual telephone

The ISDN visual telephone gateway realizes interconnection between the audio-video realtime communication adapter and ISDN visual telephone based on ITU-T Recommendations. The

gateway converts communication protocol and video-audio data frame format only, because these both equipments implement ITU-T Rec. H.261 video coding scheme and G.728 audio coding scheme. However when an ISDN visual telephone has only G.711 audio coding scheme, the gateway transcodes these coding schemes mutually.

3. Communication protocols and congestion control

3.1 Communication protocols

Figure 3 shows a call setup and release sequence over LAN. TCP protocol is used for these connection establishment sequence, and UDP datagram transmission is used for realtime transmission of coded audio and video data.

(1) Call setup

The adapter initiates a call when the adapter gets 'call' command with IP address parameter from PC/WS. The adapter transmits call setup commands to a destination adapter in TCP. The destination adapter responds with call setup accept command when the adapter establishes the call. These commands, call setup and call setup accept, have some parameters which indicate audio data length, video encoder transmission capability, video decoder mode request, etc.

(2) Coded audio and video data transmission

The adapters send and receive audio/video coded data with UDP. The audio data and video data are combined in a packet to decrease the number of packets, and transmitted at regular intervals. This is to achieve better transmission efficiency. The packet structure is shown in Figure 4. A packet consists of a control command, a sequence number, an audio data length, a video data length, coded audio data and coded video data. The control command indicates fast update request, loop back request and, transmission and reception bit rates. The sequence number shows the packet number and is used for detecting packet loss by checking it continuity. The audio data is fixed at 0, 70, 110 or 150 bytes in a packet. The video data is filled with variable length coded data at regular intervals in a packet.

(3) Call release

The adapter releases a call when the adapter gets 'release' command from PC/WS. The adapter transmits release command to the destination adapter by TCP, then stops audio-video data transmission through UDP data packets.

3.2 Congestion control

The UDP is very simple and easy to process, so it can suit realtime data transmission. However, when LAN traffic becomes heavy, UDP packets are discarded. The adapter can detect a packet discard by checking the sequence number in the data packet, then lowers video throughput by one step automatically to relieve congestion and sets the encoder in fast update mode to clear decoded video. When congestion persists, it adjusts rate one step further. Transmission rates return

to their original speed once congestion is cleared. If some audio data is lost, it is replaced with silence data to maintain audio-video synchronization.

3.3 Interconnection with ISDN visual telephone

Figure 4 shows an interconnection sequence between LAN terminals and ISDN equipments.

(1) Call setup

When a connection is established, the gateway converts communication protocols mutually. Video and audio transmission modes of both LAN and ISDN terminals are set in this step.

(2) Audio-video communication

The gateway converts data frame format and audio coding schemes. The video codec of the adapter is based on ITU-T rec. H.261, but has no BCH forward error code scheme. The gateway adds BCH forward error codes. Since the audio codec of the adapter has only G.728, the gateway transcodes it to G.711 coding scheme if G.728 is not supported by the ISDN terminal. When a visual telephone has G.728, the gateway simply transfers the audio data. Then the gateway multiplexes coded audio and video data in H.221 frame for an ISDN visual telephone.

To avoid overflow of coded video data from the adapter to the ISDN terminal, the gateway sets video data transmission rate of the LAN terminal lower than the one for LAN-to-LAN communication. For example, if 2B is the ISDN rate with G.728, LAN video rate is set to 77 kbps. When video buffer is in under flow, fill bits are buried in BCH frames.

4. Design concept and terminals

Hardware architecture of audio-video real-time communication adapter and ISDN visual telephone gateway are described here.

4.1 Audio-video real-time communication adapter

Figure 5 shows a block diagram of the adapter. LAN IF has 10 Mbps CSMA/CD LAN (ISO 802-3 10BASE5) interface, treats TCP/IP protocol, and communicates control data in TCP and coded audio-video data in UDP. PAD packs coded audio and video data in a packet. VIDEO CODEC encodes and decodes video signals based on ITU-T Rec. H.261. However, the video codec has no BCH forward error correction framing pattern. This is because the UDP packet has checksum and detects bit error.

Frame rate is 15 f/s in CIF and 30 f/s in QCIF. Data rate is 192 kbps at maximum. AUDIO CODEC encodes and decodes audio signal based according to ITU-T Rec. G.728. Data rate is 16 kbps fixed.

Audio-video real-time communication adapter is shown in Figure 6, and the specifications are summarized in table 1.

4.2 ISDN video-phone gateway

Figure 7 shows a block diagram of the gateway. LAN IF has 10 Mbps CSMA/CD LAN (ISO 802-3 10BASE5) interface, and communicates with an audio-video real-time communication adapter. PAD packs/separates coded audio and video data in/from a packet. ACC transcodes audio data when a visual telephone has only G.711 audio codec. When a visual telephone has G.728 audio codec, the ACC transfers it. The video codec of the adapter has no BCH forward error code scheme, ECF codes and decodes BCH. MUX multiplexes/demultiplexes coded audio and video data in/from H.221 frame. NIF has an interface of basic-rate ISDN circuits, and communicates with an ISDN visual telephone.

Table 2 summarizes the specifications of the gateway.

5. Conclusion

Information on an implementation of video and audio communication system for LAN has been provided to facilitate making a standardization program.

Table 1. Specification of audio-video real-time communication adopter

Item		Specification
LAN interface	LAN IF	10 Mbps CSMA/CD LAN (ISO 802.3 10BASE5)
	Comm. protocol	TCP/IP protocol
	Audio-video data packet	audio-video combined 70-1430 bytes/packet 13-29 packets/sec
	Transmission rate	16-208 kbps
Video interface	Coding algorithm	ITU-T Rec. H.261 (w/o BCH frame)
	Resolution	352x288 pixels (CIF) 176x144 pixels (QCIF)
	Data rate	16, 37, 77, 128, 192 kbps
	Input/Output signal	NTSC composite
Audio interface	Coding algorithm	ITU-T Rec. G.728
	Data rate	16 kbps
	Input/Output signal	Analog
Control interface	Interface	RS-232C
	Transmission rate	1.2, 2.4, 4.8, 9.6 kbps
Size		110(W)x340(H)x400(D) mm

Table 2. Specification of ISDN visual telephone gateway

Item		Specification
LAN interface	LAN IF	10 Mbps CSMA/CD LAN (ISO 802.3 10BASE5)
	Comm. protocol	TCP/IP protocol
	Data format	UDP datagram packet
	Transmission rate	16-208 kbps
ISDN interface	ISDN IF	Basic-rate IF (2xB)
	Comm. protocol	ITU-T Rec. H.242
	Data format	ITU-T Rec. H.221
	Transmission rate	64, 128 kbps
Control interface	Interface	RS-232C
	Transmission rate	1.2, 2.4, 4.8, 9.6 kbps
Size		110(W)x340(H)x400(D) mm

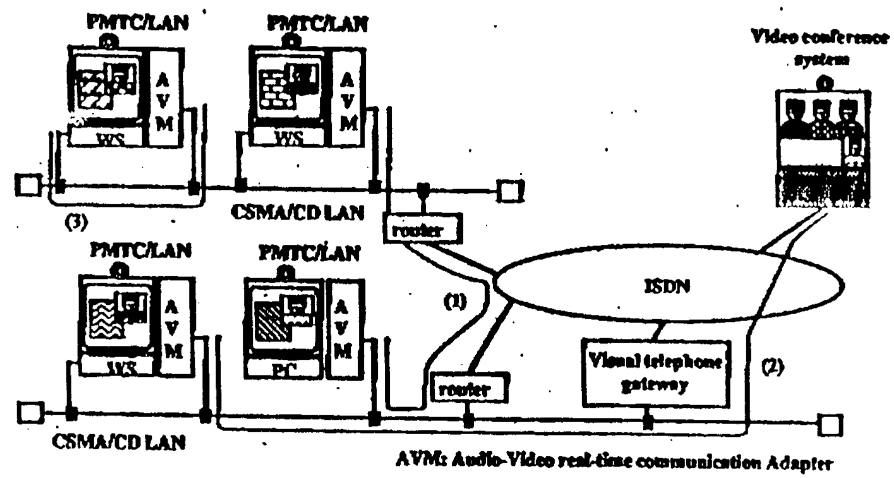


Fig.1 Multimedia Communication System for CSMA/CD LAN

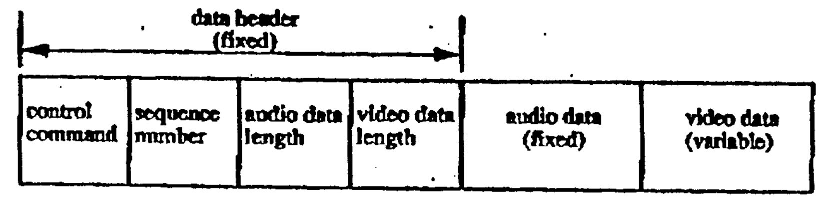


Figure 3. Audio-video data packet structure

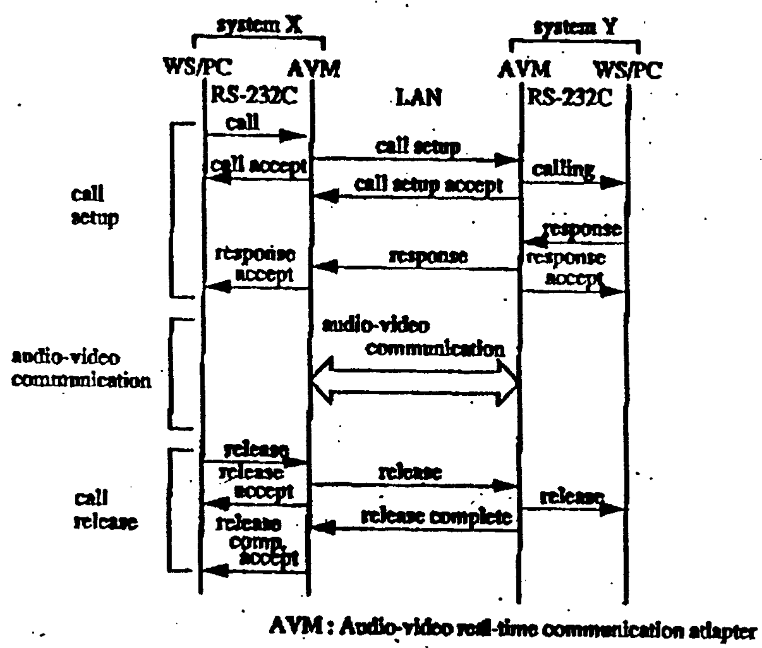


Figure 2. Call setup and release sequence

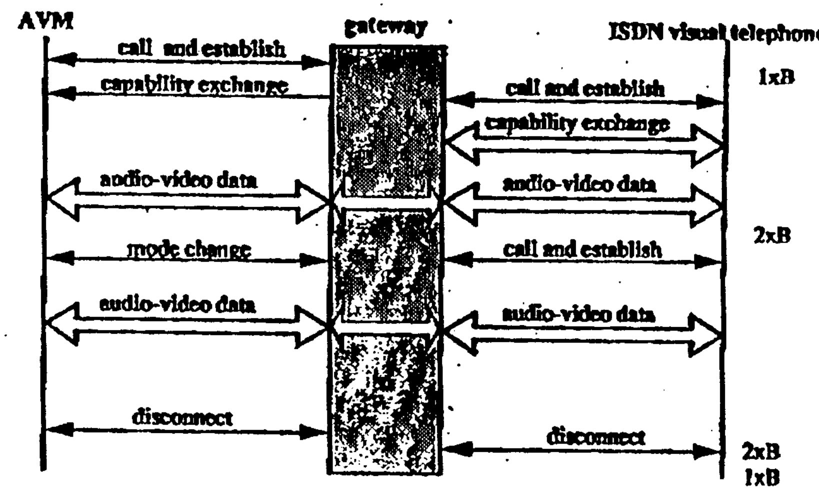


Figure 4. Call setup and release sequence

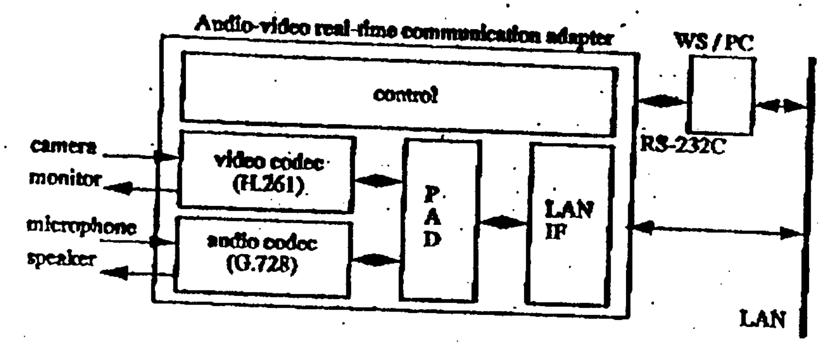


Figure 5. Block diagram of an audio-video real-time communication adapter

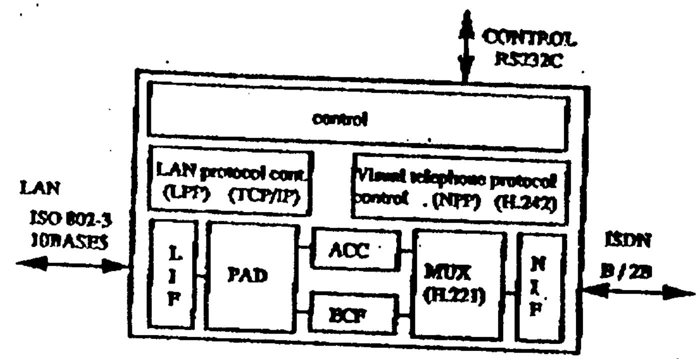


Figure 7. Block diagram of visual telephone gateway

Figure 6. An external view of audio-video real-time communication adapter

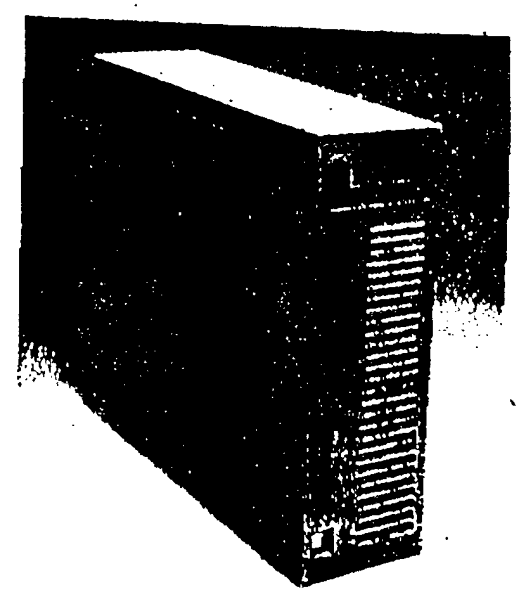


Figure 8. An external view of audio-video communication gateway

APPENDIX 5.1

SUMMARY OF THE FPLMTS PROJECT

ATTACHMENT 1

2

Documents
CCIR Study Groups
Period 1990-1994

Document 8-1/TEMP/47(Rev.1)-E
Palermo, 21 October 1992
English only

Program Management Team

Program for the introduction of FPLMTS

1. Purpose

The FPLMTS Program identifies short and long term milestones, in the form of specific output documents, the time scale for their production and the allocation of responsibilities.

2. High level program

A diagrammatic representation of the conceptual high level program is contained in Figure 1 with a detailed description of each element provided in Table 1.

It should be noted that the concepts may need to be applied differently on a per issue basis, for example, in some cases, particular elements may not be applicable and hence may be bypassed.

3. Milestones for the overall definition of FPLMTS

The milestones for the overall definition of FPLMTS have been given in annex 1. These milestones are identified from a top-down perspective with the view to get FPLMTS in operation around the year 2000. The milestones are given irrespective of responsibilities within the ITU.

4. Framework of detailed recommendations for FPLMTS

The framework of detailed recommendations for FPLMTS is given in annex 2. The detailed recommendations are the complete and detailed description of all aspects of FPLMTS that are necessary to get FPLMTS in operation. Baseline recommendations needed in the interim to arrive at specific decisions are not included in this framework.

It should be noted that the framework of detailed recommendations in annex 2 does not identify explicit CCIR or CCITT recommendations, but is more to be seen as an overview of the various technical issues that need to be addressed in order to arrive at a detailed description of FPLMTS, to be used for FPLMTS program management. It should also be noted that the responsibilities shown are based on the organizational structure of the CCIR and CCITT as of October 1992.

The detailed structure of CCIR or CCITT recommendations explicitly on FPLMTS or including FPLMTS issues will have to be defined by the parties responsible for their production. Further, it should be noted that although an identified technical issue needs to be addressed, there are no assumptions implied concerning the need for standardization or the degree of standardization for the particular technical issue.

3

Figure 1: FPLMTS Program (Conceptual)

1992	(0)	Concept Recommendation	REC 687
....	(1)	Framework and Requirements Recommendations	REC 816 Services REC 817 Architecture REC 818 Satellite REC 819 Developing et al.
End 1993....	(2)	Selection Procedures Recommendations	
End 1995.....	(3)	Key Choices Recommendations	
1997	(4)	Detailed Recommendations	
1997.5........	(5)	Standards	
....	(6)	Validated Standards	
....	(7)	Prototype	
....	(8)	Type Approval	
....	(9)	Manufacture	
2000......	(10)	System Operation	

ITU Activities

5

Annex 1

Milestones for the overall definition of FPLMTS

First overall system objectives set (CCIR Rec 687):	Mid 90
Initial framework of services defined (CCIR Rec 816):	Mid 92
Network architecture defined (CCIR Rec 817):	Mid 92
Objectives for satellite integration set (CCIR Rec 818):	Mid 92
Objectives for support of developing countries set (CCIR Rec 819):	Mid 92
Spectrum identified by WARC-92	Mid 92
Initial radio interface framework defined:	Mid 93
Initial FPLMTS spectrum usage principles defined:	Mid 93
FPLMTS traffic requirements defined:	End 93
System objectives and framework finalized:	Mid 94
Framework of services and facilities finalized:	Mid 94
Security principles defined:	Mid 94
Initial FPLMTS terminology defined:	Mid 94
Network procedures finalized:	End 94
Radio interface signalling requirements defined:	End 94
Network interworking scenarios defined:	End 94
Framework for satellite integration defined:	End 94
Network management principles defined:	End 94
Choice of radio access principles:	Mid 95
Choice of speech/channel coding principles:	Mid 95
Data services principles defined:	Mid 95
Detailed security requirements defined:	Mid 95
Initial performance requirements defined:	Mid 95
Speech and channel coding issues defined:	End 95
Physical radio access structure ready:	Mid 96
Radio interface protocols ready:	End 96
Complete audio aspects ready:	End 96
Security algorithms ready:	End 96
Video coding issues defined:	End 96
Network protocols ready:	Mid 97
Data services issues ready:	Mid 97
Detailed network management requirements ready:	Mid 97
FPLMTS terminology finalized:	Mid 97
Physical radio access performance ready:	End 97
Complete video aspects ready:	End 97
Conformance specifications ready:	End 98
Possible start of service:	2000 - 2002

7

FPLMTS technical area:	Responsibility:	To be finalized:
RADIO ASPECTS (physical layer and signalling)		
Framework of radio system	TG 8/1	12/95
Multiplexing and multiple access	TG 8/1	06/96
Channel coding	TG 8/1	06/96
Modulation	TG 8/1	06/96
Transmission and reception	TG 8/1	12/97
Physical link control	TG 8/1	12/97
Synchronization	TG 8/1	06/96
Radio interface protocols, layer 1	TG 8/1 and CCITT SGXI	12/96
Radio interface protocols, layer 2	TG 8/1 and CCITT SGXI	12/96
Radio interface protocols, layer 3	TG 8/1 and CCITT SGXI	12/96
NETWORK MANAGEMENT ASPECTS (requirements, models and procedures)		
Overall TMN framework	TG 8/1 and CCITT SGIV	12/94
Performance management	TG 8/1 and CCITT SGIV	06/97
Fault management	TG 8/1 and CCITT SGIV	06/97
Configuration management	TG 8/1 and CCITT SGIV	06/97
Accounting management	TG 8/1 and CCITT SGIV	06/97
Security management	TG 8/1 and CCITT SGIV	06/97
End user profile management	TG 8/1 and CCITT SGIV	06/97
AUDIO ASPECTS (detailed descriptions)		
Speech codecs description	CCITT SGXV	12/95
Voice Activity Mechanisms	CCITT SGXV	12/96
Echo control	CCITT SGXII	12/96
Transmission requirements	CCITT SGXII	12/96
VIDEO ASPECTS (detailed descriptions)		
Video codecs description	?	12/97
Transmission requirements	CCITT SGXII	12/97
TERMINAL ASPECTS (functional requirements)		
Terminal adaptation functions	?	06/97
SECURITY ASPECTS (principles, functions and detailed descriptions)		
Security for FPLMTS	TG 8/1 and CCITT SGXI	06/95
Security algorithms for FPLMTS	-- TG 8/1 or national	12/96

9

ATTACHMENT 2

Documents
CCIR Study Groups
Period 1990-1994

Document 8-1/TEMP/66(Rev.1)-E
Palermo, 22 October 1992
English only

Task Group 8/1

DRAFT OPINION

The CCIR,

considering

a) that the CCIR has a program on Future Public Land Mobile Telecommunication systems (FPLMTS) which would enable worldwide compatibility;

b) that major programs for future mobile communications within each Region are at early stages;

c) that resources of budget, manpower, and planning expertise are available to these programs which substantially exceed those readily available to the CCIR;

d) that, without international coordination, these regional programs would tend to diverge;

e) that international standards for future mobile communications (i.e. FPLMTS) will not be effective unless these regional programs are harmonized;

f) that the production of CCIR Recommendations on FPLMTS will be an important step in achieving this harmonization,

is of the opinion

that the ITU, as a matter of policy, should make every effort to persuade regional and national authorities to support the CCIR in an explicit manner in its development of Recommendations on FPLMTS and strongly encourage regional organizations to work together towards a single worldwide standard.

APPENDIX 5.2

RELATIONSHIP BETWEEN FPLMTS AND UPT

Q.7,12/15

TD 12(15-2)

LBC-93-088

INTERNATIONAL TELECOMMUNICATION UNION

RADIOCOMMUNICATION
STUDY GROUPS

Document 8-1/TEMP/106-E
Montpellier, 10 June 1993
English only

Source: Document 8-1/231

TG 8/1 Working Group III

DOCUMENT CATEGORY	: Working Document
SOURCE	: Working Group III
STATUS	: Working
DATE	: 10 June 1993
TO	: Participants to Task Group 8/1
ACTION	: Comments from participants
SCHEDULE	: By the next TG 8/1 meeting (October 1993)

Working Document

CONSIDERATIONS ON THE RELATIONSHIP BETWEEN FPLMTS AND UPT

1. General

UPT is a service concept that is provided by a UPT service provider which in principle may be different from the FPLMTS service provider. This does not preclude that a FPLMTS service provider also may be a UPT service provider, and vice versa.

FPLMTS supports UPT, and thus provides mobile access to UPT users, which logically are different from FPLMTS users. A UPT user is a FPLMTS user by default when accessing FPLMTS, but does not necessarily have to be associated with a regular FPLMTS subscriber. Conversely, a FPLMTS user does not need to be a UPT user.

2. Mobility and Portability Concepts

Simply by being a radio system, FPLMTS will have the possibility to provide "terminal mobility". In addition, FPLMTS supports UPT with the inherent "personal mobility" offered by UPT.

Further, FPLMTS should have the possibility for connecting standard Terminal Equipments (TEs) to FPLMTS Mobile Terminations (MTs), thus providing "standard terminal portability". A FPLMTS Mobile Termination is the part of the FPLMTS Mobile Station which terminates the radio path at the mobile side and adapts the capabilities of the radio path to the capabilities of the terminal equipment.

An additional feature which is being considered for FPLMTS is the "FPLMTS user mobility". The FPLMTS user mobility is a feature offered by FPLMTS simply because the FPLMTS user, and its associated FPLMTS subscription, may be physically separated from the FPLMTS terminal (e.g. with some form of a smart-card).

There are various advantages in physically separating the FPLMTS user from the FPLMTS terminal, including:

* The FPLMTS user will have some form of discrete mobility between Mobile Stations, the "FPLMTS user mobility".
* The security involved in the FPLMTS services are substantially improved, for the FPLMTS users and subscribers as well as the FPLMTS operators and service providers.
* There is much greater flexibility for the FPLMTS services providers in handling the FPLMTS users, and much greater flexibility for the FPLMTS users/subscribers to change FPLMTS service providers.
* The FPLMTS Mobile Station market will be more open since there are no requirements for associating a Mobile Station physically to a subscription.

3. Numbering, Identification and Addressing

It may be appropriate to separate the identification of FPLMTS users and the FPLMTS Mobile Terminations. This gives the FPLMTS networks the possibility to address FPLMTS users and FPLMTS terminals independently, as appropriate, depending on the choices of the FPLMTS subscriber or service provider. It also simplifies the handling of UPT users accessing the FPLMTS network, as the UPT user identity simply may be associated directly with the FPLMTS Mobile Termination identity.

The FPLMTS users and the FPLMTS Mobile Terminations are always logically separated, but there is still the option to have the two physically integrated into one equipment, if this is desired.

Concerning numbering for FPLMTS, a FPLMTS user will always have a dialable "FPLMTS Number", with which the calling parties can reach the FPLMTS users. A FPLMTS number may be mapped on to a FPLMTS Mobile Termination identity or to a FPLMTS user identity, depending on the agreements set up at subscription time between the FPLMTS service provider and subscriber.

One objective of FPLMTS is that it should support Universal Personal Telecommunication (UPT). This means that a UPT user can access service via the FPLMTS network, by using his "UPT Number". There is, however, no requirement that a FPLMTS user needs to have a UPT number (or, in other works, needs to be a UPT user). If, however, a FPLMTS user also is a UPT user, he may be reached by calling parties via FPLMTS by the use of his FPLMTS Number and his UPT Number, as appropriate.

The FPLMTS Number, whether it is used for FPLMTS Mobile

Terminations or for FPLMTS users, will have approximately the same requirements on the numbering plan as the UPT Number. Therefore, the numbering plan appropriate for the FPLMTS Number is also CCITT recommendation E.168.
The mobility and portability concepts involved with FPLMTS and UPT are illustrated in fig 1, together with the numbers and identities foreseen.

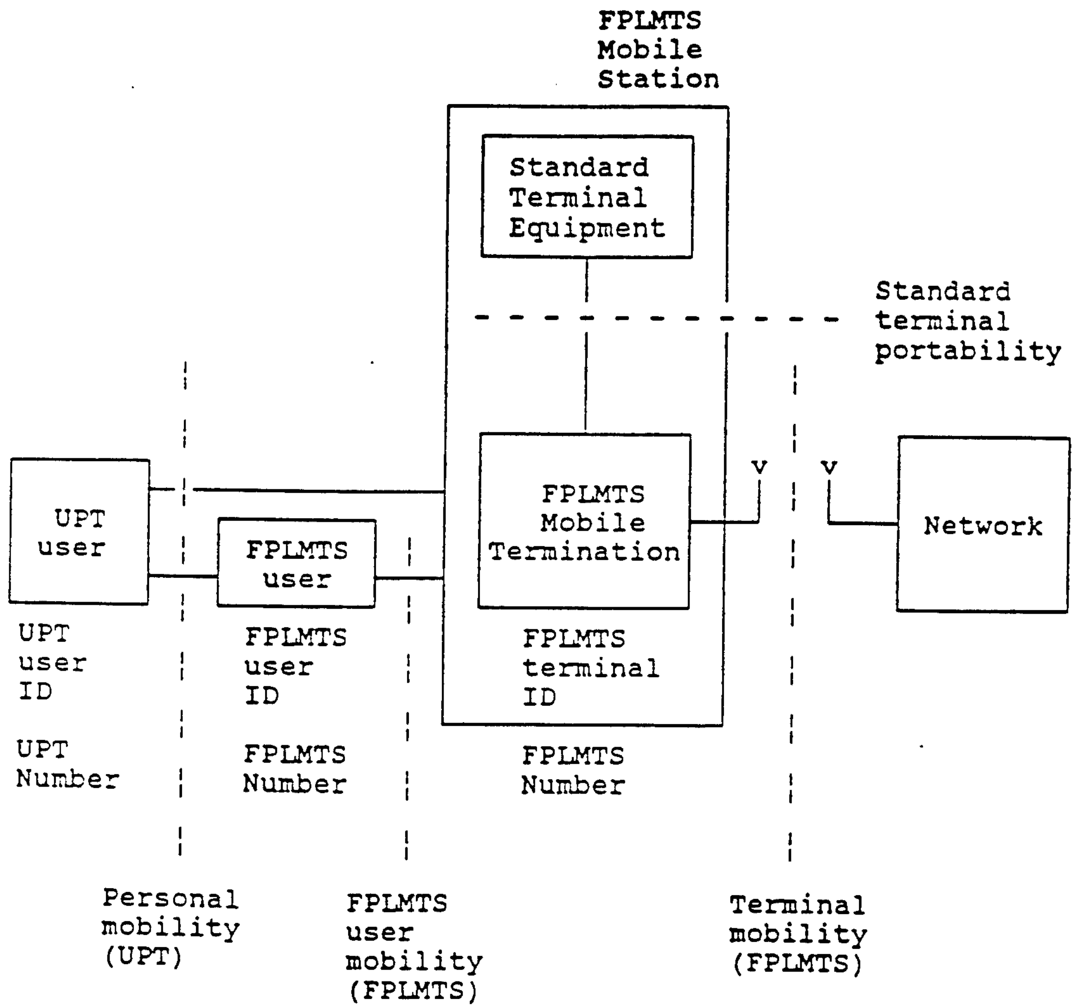


Fig 1. FPLMTS and UPT mobility/portability concepts and related numbers and identities.

4. Agreements reached at June 1993 TG 8/1 meeting

The following agreements were reached at the June 1993 joint meeting of WG III, WG IV and other interested TG 8/1 participants. Text based on these agreements is required for inclusion in this working document.

a) Subscription to UPT and FPLMTS are independent, i.e., an FPLMTS user may optionally subscribe to UPT and a UPT user may or may not be an FPLMTS user.

b) The terminal identity and user identity are logically separated. They may optionally be physically separated, thereby providing FPLMTS user mobility.

c) FPLMTS will require a diallable number (FPLMTS number). This number should conform to E.164 with characteristics similar to UPT (E.168). FPLMTS numbers will be associated with FPLMTS user IDs (in the case of FPLMTS user mobility) or with the FPLMTS terminal (in the case of the user ID logical entity being physically combined with the terminal ID entity in the terminal). This matter needs to be further considered by TS SG 2.

d) FPLMTS subscription is associated with an FPLMTS user, not an FPLMTS terminal

ANNEX 1

Vocabulary

* FPLMTS user: A person, entity or process actually using the FPLMTS services. A FPLMTS user is associated with a unique FPLMTS user identity.

* FPLMTS subscriber: A legal person or entity associated with the FPLMTS subscription and responsible for the charges incurred by his associated FPLMTS users, if any. A FPLMTS subscriber may be responsible for several FPLMTS users.

* FPLMTS service provider: A legal person or entity responsible for providing FPLMTS subscriptions to FPLMTS subscribers.

* FPLMTS operator: A legal person or entity ultimately responsible for providing complete FPLMTS network functionality to FPLMTS users. Parts of the complete FPLMTS network functionality may, however, be provided by other parties.

* UPT user: A user using UPT services, and which is associated with a UPT subscriber and UPT service provider.

* UPT subscriber: The subscriber associated with a UPT user. A UPT subscriber subscribes to a UPT service provider.

* UPT service provider: A legal person or entity responsible for providing UPT subscriptions to UPT subscribers.

APPENDIX 5.3

GUIDANCE FOR THE WORK OF THE RAPPORTEUR FOR THE VIDEOPHONE PART OF QUESTION 2/15

ANNEX B

ITU-TS

Temporary Document TD_ (15/1)

STUDY GROUP 15
WORKING PARTY 15/1
Geneva, 6-17 September ,1993

Title: Guidance for the Work of the Rapporteur for the Videophone (Very Low Bitrate) Part of Question 2/15

Source: Chairman of Working Party 15/1

1.0 Introduction

At a meeting on 6-17 September 1993, Working Party 15/1 re-appointed a Rapporteur for Very Low Bitrate Visual Telephony. The purpose of this document is to define the guidelines, requirements, objectives and work method for the work of the Rapporteur.

The Rapporteur presented a report (TD 7) at the 6-17 September meeting outlining work accomplished since the last meeting, tentative technical conclusions, and a recommended work plan. This report may be used as general guidance for the Rapporteur's work, subject to the particular requirements and objectives outlined below.

2.0 Objectives

- The objective is to develop two sets of draft ITU-TS Recommendations for Very Low Bitrate Visual Telephony. The first, (H.VLC/N), employing near term technology would be finalized in 1995. The second, Recommendation (H.VLC/L) employing more advanced technology, would be finalized in approximately 1998.

  H.VLC/N will consist of a number of Recommendations for major functional elements of the videophone system such as those noted below:

  -A Videophone System Operating at a Very Low Bitrate (H.32P)
  -Video Coder for The Very Low Bitrate Videophone (AV.268)
  -Speech Coder for The Very Low Bitrate Videophone
  -Multiplex/Error Control for The Very Low Bitrate Videophone (H.22P)
  -Supervisory Control for The Very Low Bitrate Videophone (H.42P)
  -Data Interface for The Very Low Bitrate Videophone
  -PSTN Modem for The Very Low Bitrate Videophone (V.32bis, V.34/V.8{V.FAST})

H.VLC/L will include additional Recommendations in technical areas requiring more time to develop such as:

- Advanced Video Coding
- Advanced Speech Coding
- Operation Over The Future Public Land Mobile Telecommunications System (FPLMTS)

o H.VLCN must be prepared for the future and must pave the way for H.VLC/L in such a way that the transition from the near term to the long term standard will be relatively easy. Backward compatibility is required.

o Follow the guidance from SG 1 outlined in their TD 27 Liason Statement. (Annex A)

o Follow the guidance from SQEG outlined in TD 28. (ANNEX B)

o As an objective, an optional data channel would be included to be multiplexed with the audio and video signals. Provision for high resolution still images using the JPEG standard will be provided. It is a goal to interwork with other related ITU Recommendations.

o The speech coder objective for H.VLC/N is to achieve as near toll quality as possible given the bit-rate budget. In long term H.VLC/L it is expected to achieve toll quality at 4kbps. This work has been referred to the speech experts within Working Party 15/2

o The objective for H.VLC/N is to achieve a picture quality significantly better than H.261 when operating with the corresponding parameters.

o The objective for H.VLC/L is to achieve picture quality considerably better than H.VLC/N.

o Full audio mixing in multipoint

3.0 Requirements

o ~~Consideration~~ Cater for multi-point operation.

o A flexible, robust multiplex structure to maximize the utility of the available transmission bit rate.

o Use of V.32bis and V.FAST modem technology for H.VLC/N to maximize the transmission bitrate while providing adequate error resilience.

o Interoperability with H.320 terminals

4.0 Work Method

1. In order to achieve good results, the Rapporteur will convene experts wishing to contribute to the work.

2. Work should be accomplished through correspondence as much as possible.

3. The Meeting of Experts between meetings of the WP 15/1 must be approved by the WP 15/1.

4. The Rapporteur must coordinate the work with other Study Groups and other appropriate standards bodies. Study Group 15 will transmit official requests for cooperation to other Standards Groups and Standards Bodies when required.

5. Work jointly with SG 1 to develop a detailed set of Technical Requirements for all functional elements of the Very Low Bitrate Videophone.

6. Collaborate with ISO/IEC JTC 1/SC29 WG11 (MPEG4), particularly in the area of advanced video coding.

7. The Rapporteur will provide progress reports at all meetings of Study Group 15 and/or Working Party 15/1.

APPENDIX 5.4

VIDEO CODING IN MOBILE NETWORKS - SOME ASPECTS

TSS SG15 Special Rapporteur
for very Low Rate Visual Telephony

LBC-93-070
Boston, 9.-10. July 1993

Title : Video Coding in Mobile Networks – Some Aspects
Source : BOSCH C/FOH and IENT Aachen [1]
Purpose : Discussion

1 Introduction

Mobile telecommunication is of increasing importance. This point of view is justified by the development of the European *Universal mobile telecommunication System (UMTS)* [1] headed by ETSI STC SMG5. Comparable work is performed internationally by ITU-R TG8-1 responsible for the standardization of the *Future Public Land Mobile Telecommunication Systems (FPLMTS)* [2]. These networks will be used as a basis for all mobile telecommunication services by the year 2000. The proposed range of applications can be described as multimedia services. Thus within these networks video transmission will be only one possible service. One important project in this context is the European RACE project MAVT R2072 currently developing a first mobile audio-video terminal [3].

The picture quality of future video services must be significantly better than the quality of current codecs. Thus high sophisticated video coding algorithms are necessary to meet the demand for very high quality. This target is very close to that of MPEG-4. so MPEG-4 can be seen as a possible video coding standard used within the UMTS (FPLMTS). Further the time schedule is well adjusted. As a result MPEG-4 chips will be the first choice for UMTS.

Since there are basic differences between mobile networks and fixed networks, mobile aspects should be considered in the development of novel video coding algorithms in MPEG-4 from the beginning.

2 Mobile Aspects

There are mainly two differences between mobile and stationary video communication:

- image material
- transmission aspects

[1] contact D.Lappe or Kl.Illgner

2.1 Scenes in a Mobile Environment

In mobile environments there are no restrictions of the scenes to be transmitted to some purposes. The scenes are much more complex than those currently assumed in fixed networks at very low it rates, e.g. Miss America or Salesman. The characteristics of the image material can be summarized as follows:

moving camera : zoom, pan, vibration, auto focus

motion types :

- strong movements possible
- motion in the background [2]
- much stronger motion in the 3D space compared to video conference like scenes

Thus there are a lot of regions with highly varying motion.

objet types : arbitrary objects, highly structured background

scene cuts : or scene cut like situations characterized by very fast pan and/or rapidly changing background

outdoor scenes : illumination changes, varying contrast, whether conditions as rain, mist, snow, etc.

noise : camera noise

2.2 Transmission Aspects

In fixed telecommunication networks the error probability on the transmission links is very small ($P_e \approx 10^{-7}$). Thus using error correcting codes an almost error free data transmission is possible.

However in mobile channels an error free data transmission can not be guaranteed. Thus the coder must consider the current channel characteristics to enable a reliable data transmission. As a result the coder is influenced by the channel.

[2] What should be the definition of background if no dominant object is present?

In the following some important aspects are listed :

channel characteristics :
time and/or frequency selective fading, burst errors

channel capacity :

- only small channel bandwidth
- Due to the varying error probability a stationary or even fixed net channel capacity can not be assumed.

several channels :
To ensure a target picture quality more than one channel might be necessary. This introduces additional problems as how to distribute the information over the channels and to achieve a stable transmission.

bit error sensitivity :
Some types of symbols are more sensitive to bit errors than others. More significant information needs a higher error protection than less significant information [4].

synchronization :

- As a result of not correctable errors coder and decoder will lose synchronization.
- There might be a loss of data bits e.g. in case of handover.

3 Prerequisites for Video Coding Algorithms

The described characteristics of the image material and the transmission lead to conditions a video coding algorithm should fulfill.

- Concerning the image material model-based approaches assuming special types of scenes as e.g. head and shoulder scenes might not be the best solution.
- Complex motion models are necessary to cope e.g. with different overlaid types of motion.
- Scene cuts and similar situations must be detectable, because temporal prediction might fail.
- Due to the varying net channel capacity the coding algorithm should be able to generate different bit rates as close as possible at the rate-distortion bound for each generated rate.

- To meet the requirement of high quality even if the net channel capacity is low (high error probability) the coder should be able to use more than one channel. A multi-resolution approach might be a solution with a very high error protection in the base layer to avoid losing significant information necessary for decoding the additional layers.
- The selection of the information to be coded might consider the dependency between different types of symbols; e.g. in current algorithms if there in an error in a motion vector the DCT-coefficients for that block are useless.
- It can be assumed that there will be some status information from the decoder available at the coder. [3]
- A coder – decoder resynchronization is necessary. That means that the type of coded information may be influenced by the decoder!

4 Conclusion

The increasing importance of mobile networks (UMTS) requires the consideration in the development of novel video coding algorithms. The close relation between the topics of MPEG-4 and UMTS seems to favour the coming MPEG-4 video coding algorithm for usage in UMTS.

The paper tried to point out some important aspects of mobile networks and to give guidelines for the development of video coding algorithms useful in these networks.

5 Acknowledgement

The paper was written as part of a joint investigation project with Robert BOSCH GmbH, Research Institute Hildesheim, Germany. This work has been partly founded by the European Comission under the contract RACE R2072 MAVT.

[3] But what should be done in case of a multipoint link?

References

[1] Stein Hansen. *UMTS Standardization.* Norwegian Telecom Research, RACE Mobile Workshop, Metz, June 1993.

[2] ITU Radiocommunication Study. *Spectrum Considerations for Implementation of F-PLMTS.* ITU-R 8/1, 1993. Q.39/8.

[3] D. Lappe, S. Panis, F. Pereira, M. Roser, K. Rijkse, and D. Thoreau. *First Reference Model for Video Coding.* RACE R2072 MAVT. R2072/BOS/2.1/DS/R/005/b1.

[4] R. Mann Pelz. R2072 MAVT: *Channel coding aspects.* to be published in RACE R2072, June 1993.

APPENDIX 5.5

PROPOSAL FOR A GENERIC DATA STREAM STRUCTURE CONSIDERING CURRENT SHORT-TERM AND LONG-TERM STANDARDIZATION ACTIVITIES

TSS SG15 Special Rapporteur
for very Low Rate Visual Telephony

LBC-93-069
Boston, 9.-10. July 1993

Title : Proposal for a Generic Data Stream Structure Considering Current Short–term and Long–term Standardization Activities

Source : BOSCH C/FOH and IENT Aachen [1]

Purpose : Discussion

1 Introduction

The current situation in the field of very low bit rate video coding shows uncertanties about the applications low bit rate coding should be used for. On the one hand there is the PSTN videophone. For this application a video coding algorithm has been developed in COST211ter [1]. But there are also devices (AT&T, Marconi) already on the market. On the other hand a mobile videotelephony hardware is currently under development in the RACE project R2072 MAVT (see related paper).

Although all coding schemes under consideration use a DCT-based hybrid DPCM loop and therefore the same type of data is coded, the data stream structures are partly or even completely incompatible (fig.1).

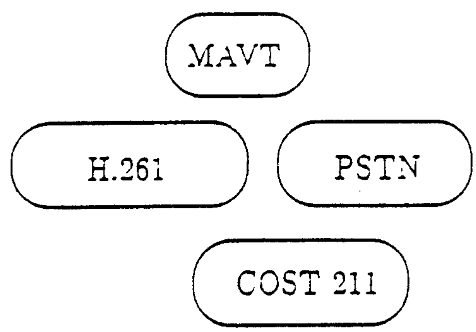


Figure 1: Compatibility of low bit rate videocoding algorithms

Thus to achieve interoperability one has to make compromises or one might need switching devices as e.g. in ISDN - PSTN connections. These solutions can introduce serious problems in the future, because there is a fast development of services and demands in the area of telecommunication. Additionally the rapid development in the area of mobile

[1] contact D.Lappe or Kl.Illgner

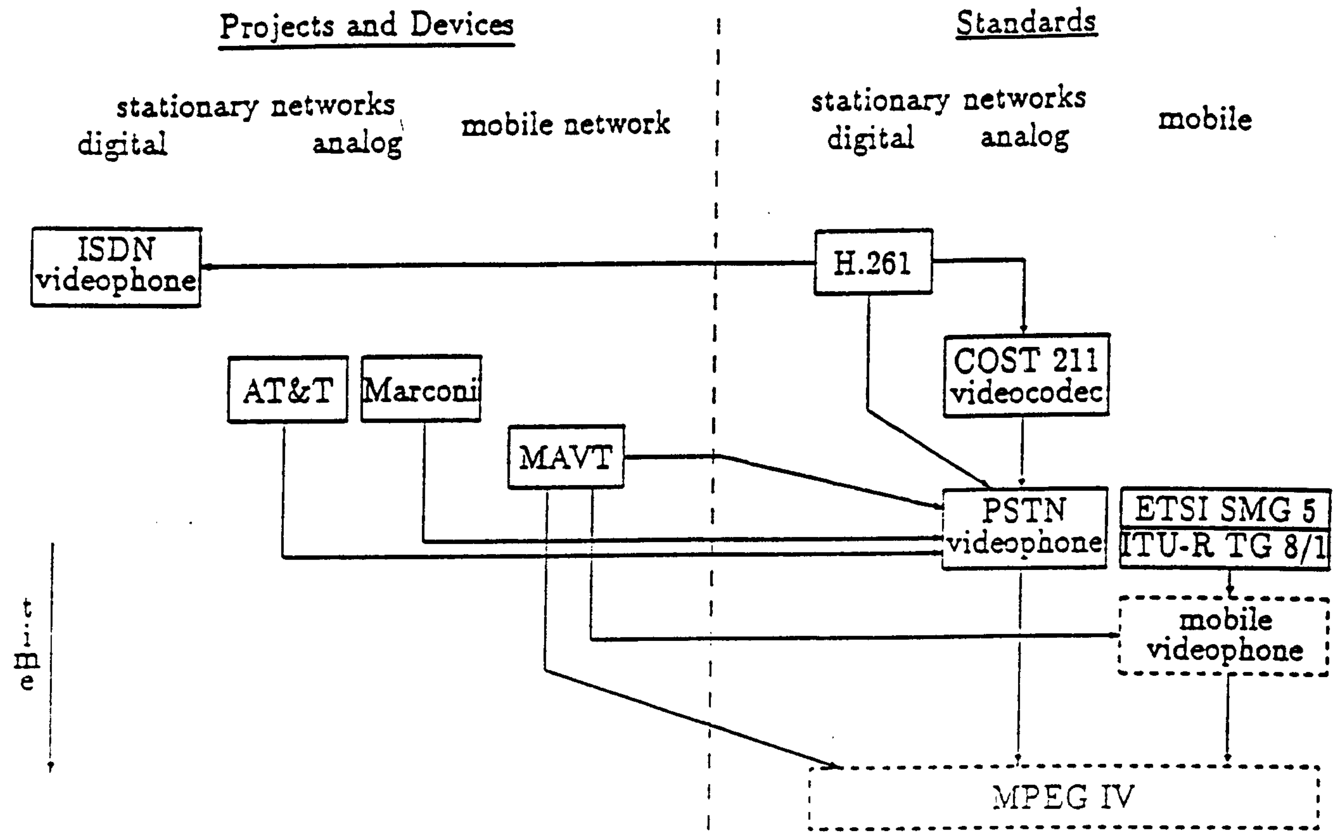


Figure 2: Evolution of low bit rate videocoding standards and devices

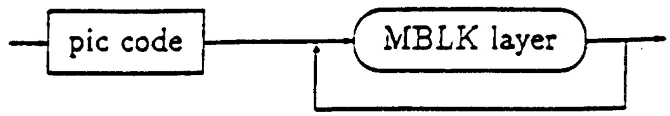
telecommunication (UMTS) should be considered. These problems can be avoided by defining more flexible standards assuming one type of applications as e.g. video coding via point-to-point links. The main intention should be that encoder and decoder agree upon coding parameters and transmit the coded data with a generic structure.

One step towards more flexible video coding standards is the definition of a generic data stream structure. It should have the capability to include all known data structures and must consider specific problems of target applications as e.g. mobile communication. For future demands the data structure in question should additionally allow more sophisticated coding algorithms as e.g. region oriented coding schemes and hierarchical coding (fig. 2).

2 Currently used data structures

In current block based video coding algorithms the symbols to be coded are the motion vectors and the DCT-coefficients for each block or macroblock. Additionally there are overhead bits for the coder control signalling e.g. the update mode (INTER/INTRA). There are two main possibilities for multiplexing of these symbols into one video data

picture layer:



MBLK layer:

CMP	VLC	mode information and description which blocks are coded
motion	VLC	motion vectors of the macroblock, horizontally predicted
DCT	VLC	quantized DCT-coefficients of the coded blocks (see CBP)

Figure 3: Multiplexing of the symbols in the H.261

stream.

2.1 Sequential Multiplexing

This scheme is used in the H.261 [2] and in the COST 211ter Simulation model [1]. All data for one macroblock is collected and transmitted. In the so-called macroblock layer (MBLK layer) first some codewords signal what has been coded for that macroblock. In the second part the motion vectors followed by the DCT-coefficients are transmitted. The macroblocks are processed sequentially (fig. 3).

An advantage is that coding and decoding may be done just in time introducing only small coding delay. Another advantage could be less storage requirements. The disadvantage of this concept is that an adaptation to global characteristics is hardly possible. Each macroblock is handled more or less independently from the other macroblocks. Thus it is difficult to implement e.g. a region oriented scheme.

The characteristics of mobile channels is completely different compared to stationary channels. One aspect is the error probability. In the ISDN the worst case error rates are $\approx 10^{-6} \ldots 10^{-7}$ whereas in mobile channels burst errors with rates up to $\approx 10^{-1} \ldots 10^{-2}$ must be considered [4]. Thus in mobile videophones the error protection has to be adapted to the different bit error sensitivities; e.g. motion vectors are more sensitive than coded DCT-coefficients and require therefore more error protection.

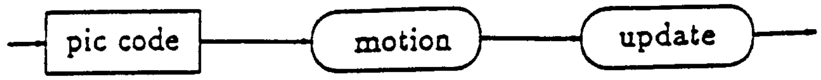


Figure 4: Multiplexing in MAVT

2.2 Global Multiplexing

In the alternate scheme which is used in the MAVT videophone (fig. 4) all data of one type is collected for the whole picture and transmitted as one block. Thus first the complete displacement vectorfield is transmitted and in the second block the update information.

This concept of data partitioning allows independent error correction on various types of data. Further a more efficient coder control is possible, because all data for coding of a complete image is available at once. So one can optimize the update information in terms of the available bitrate.

The main disadvantage is a slightly greater storage requirement and a coder delay of at least one picture.

3 Prerequisites

The definition of a generic data stream structure has to consider current data stream structures. Further the requirements for mobile environments must be taken into account.

It is expected that for mobile videophones at least two channels each with a net bitrate of 9.6 kbps are necessary to meet the quality requirements. Therefore the coding structure should be based on a *layered* scheme.

The video coding algorithm must be able to adopt to variable net bitrates on the channel. Thus the data stream structure has to consider scalable coding schemes. [2]

For future demands (MPEG IV ?) region oriented codecs should be able to use the same data stream structure.

The requirements to consider are summarized as follows:

- capability for interworking with the H.261 data stream structure (no transcoding, only reformatting)
- stability against burst errors which occur in mobile channels
- scalability - nearly optimal performance at different bitrates
- dynamic channel allocation - layered coding

[2] The terms *layered* and *scalable* coding are not well defined. Thus in the context of this paper layered coding means refinement only of the resolution using seperate data streams. Scalability mostly has the meaning of partially decodeable data streams, but stands in that paper for the generation of varying bit rates with nearly optimal performance.

- support of region oriented coding schemes
- fast picture startup mode - necessary e.g. in case of severe channel errors

4 A generic Data Stream Structure

The global multiplexing scheme is better suited for mobile environments and is more flexible to meet the prerequisites. Thus we propose as a generic data stream structure an extension of the global multiplexing used in MAVT.

Between the picture startup code and the motion information a block describing the coded regions is inserted (fig. 4,5). The idea behind this is that only regions itself and their associated motion and update information should be coded.

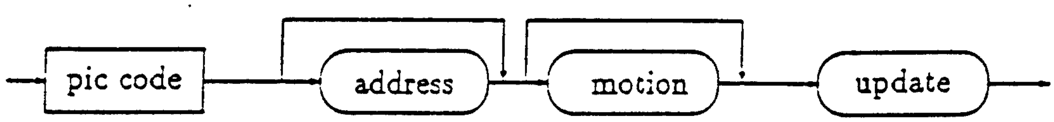


Figure 5: Extended multiplexing

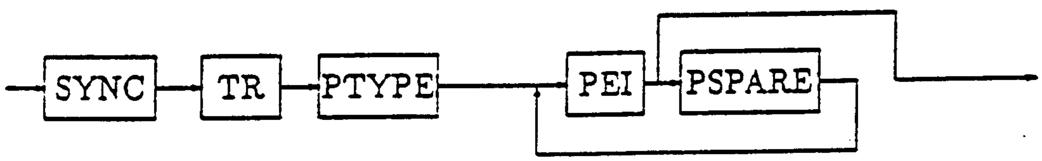
4.1 Description of the parts

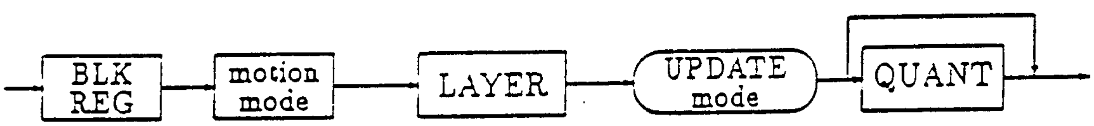
The picture startup code (fig. 6) handles general information valid for one frame, e.g. block or region oriented coding. The modes for motion compensation (e.g. displacement vectors or parametric motin decription) and update coding (e.g. DCT, subband coding) are signalled here, too. Special update information globally valid as e.g. the quantizer stepsize can also be included here. Further one could think about VLC-codewords signalling one set of modes instead of coding each mode separately.

In the address part (fig. 7) the shape and position of each coded region is transmitted. Also the update mode for that region can be signalled, if the desired mode is different to the global mode. First the position of the region to be coded is transmitted. This can be the upper left corner of a surrounding rectangle relative to the previous region position. The coding method for the shape is still open because efficient lossless contour coding schemes are very expensive compared to the available bitrate. Lossy coding via spline approximation seems to be more promising than chain codes. However this approach introduces additional problems in case of layered coding.

Block oriented coding is handled by coding the number of skipped, that means not coded, blocks assuming a sequential scanning in the position codeword. This replaces the COD bit from the COST 211ter Simulation model. Thus each block is regarded as one region. No shape information needs to be coded. If the type of motion or update mode differs from the global mode the locally used modes are signalled in the coding mode

pic code:





SYNC	20	picture startup synchronisation
TR	5	time reference
PTYPE	1	fast update or normal mode
PEI	1	signalling extra information in PSPARE
PSPARE	8	extra information
BLK/REG	2	Switch whether arbitrarily shaped regions are coded or blocks. If blocks are coded no shape information is transmitted later but one further bit signals a blocksize of 8x8 or 16x16 pel
motion mode	2	parametric motion description, motion vectorfield or e-ven no motion
LAYER	2	number of extended layer
UPDATE mode	4	DCT (always 8x8), subband coding, pyramid coding
QUANT	5	main quantizer stepsize

Figure 6: extended picture code (pic code)

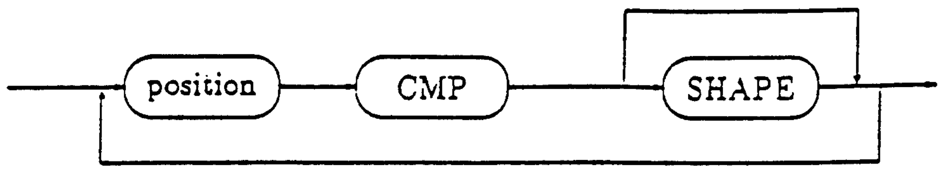
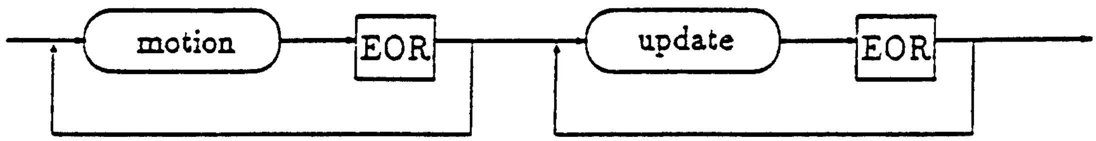


Figure 7: Multiplexing within the address part



EOR End of region

Figure 8: Multiplexing within the motion and update part

pattern (CMP) codeword. Thus one can switch within one frame between parametric motion description or displacement vectors and between DCT-coding or subband coding.

The motion information coded next may be motion parameters or a displacement vectorfield. If displacement vectors are used as motion information they are predicted only within the associated region, not across the region boundaries. In this block all motion information is stored sequentially but only for those regions for which motion information has been signalled in the address part. Additionally one could agree in case of block oriented coding upon that the motion vectors are not the original motion vectors but are predicted from the neighbored vectors (DPCM instead of PCM).

Coding of the update information only associated with one region is easily possible. The same principle for arranging the update information as for the motion information is used. No further considerations have to be made in case of transform coding. Using a hierarchical coding scheme, e.g. a pyramid, the update information must be extended by a number signalling the layer of the pyramid. This can be done by appending a second update block for the second layer just behind the first update block. However an independent layered coding as described below seems to be more suitable. Subband coding is also possible regarding one subband as one layer. Within each subband only the spatial region corresponding to the region in question is coded.

4.2 Extensions

Independent layered coding can be implemented by using the same data structure (fig. 5). The administration information and the region information is coded in the base layer. The second layer is transmitted in the following frame starting again with a pic code. The numbe of the coded layer is signalled in the picture startup code. Assuming that there

is no motion information in the second layer one can skip the region block as well as the motion block and can just transmit a second update block. The refinement information can be a more detailed region description considering the already known description. Also one can code just a second update block containing additional pyramid layers or bandpasses. But even a more detailed motion description is possible by signalling this in the "second" picture startup code.

If necessary even nearly the original data stream structure of e.g. COST can be generated by coding only the first macroblock in the "first" picture layer. The next macroblock is coded in the "second" picture layer and so on. The disadvantage of a lot of picture startup codes may be solved by using only one bit for signalling whether there is another layer or not-

The structure can be refined by introducing a sequence startup code as in MPEG [5]. In general there must be a handshake procedure at the beginning of a video session. This handshake procedure includes the transmission of mode and update information valid for one sequence. Thus there can be a reduction of bits necessary for administration.

5 Conclusion

In the paper a proposal for a generic data stream structure is described. A common and flexible data stream structure is very important because there will be a fast development of services and demands in the area of mobile telecommunications. Thus it is impossible to fix a specific data structure and simultaneously fulfill future demands of compatibility with existing devices. A flexible data stream structure can handle a broad variety of applications. The coder and decoder will finally agree upon what actually will be transmitted.

It can be expected that future mobile videocodecs contain a motion estimation and motion compensation part and a part for coding the residual image. For example the analysis-synthesis coding developed at University of Hannover [3] and introduced at COST 211ter and MPEG-4 as a starting point for the development of future coding schemes can be handled by that data stream structure.

Thus the proposed data stream structure might be useful even for the future coding scheme developed by MPEG-4. Within the project RACE MAVT this data stream structure is expected to be used.

References

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[3] M. Hötter. Object-oriented analysis-synthesis coding based on moving two-dimensional objects. *Signal Processing: Image Communication*, 2(4):409 – 428, December 1990.

[4] R. Mann pelz. R2072 - MAVT: *Channel Coding Aspects.* to be published in RACE R2072, June 1993.

[5] ISO-IEC/JTC1/SC29/WG11. *Description of the MPEG-2 Test Model*, July 1992. AVC-323.

APPENDIX 5.6

R2072: MATV -
THE MOBILE AUDIO-VISUAL TERMINAL

ITU-TS SG15 Rapporteur
For Very Low Bitrate Visual Telephony

Document LBC-93-072

R2072: MAVT - The Mobile Audio-Visual Terminal

Workplan References:

T.307, T.311, T.313, T.317, T.320

Project Line:

PL3: Mobile Communications
PL4: Image Communications

Main Objectives

The project objective is to find powerful video and audio coding algorithms for the transmission of moving and still video in a mobile environment (like UMTS), and implement them on a demonstrator. This necessitates a study of user requirements, network and channel-characteristics, service definitions and a general terminal architecture. The project will deliver future algorithms for low bitrate video (px 8k bit/s) and audio coding, and a futuristic terminal with several novel features, including a variable resolution display and good quality audio. New proposals for VLSI chips for video and audio coding are expected from realisation of the demonstrator.

Technical Approach

Following development of the px8kbit/s video coding algorithm and the low bitrate audio algorithm, the project will concentrate on research in channels for mobile video transmission. New methods like RCPC Codes and a flexible exchange between source and channel data rates will be analysed, as well as combined source and channel coding methods. Demonstrator hardware will be built within the DECT environment, consisting of multi-processor DSPs and VLSIs. For the UMTS environment a full transmission simulation will be done.

Key Issues

- Service definition and user requirements.
- Influence of network and channel characteristic.
- Suitable video and audio coding algorithms.
- Channel error protection for video and audio.
- System control.
- Demonstrator realisation with modern DSP and VLSI circuits.

Achievements

MAVT has identified possible video and audio services for a mobile terminal, and has defined the input video formats for those services. A reference model for low bitrate video coding was created, and a first algorithm for px 8kbit/s video coding was produced.

New algorithms were developed that optimise low bitrate video and audio coding in terms of bitrate, complexity and transmission delay. Some are based on current standards (H.261, MPEG, JPEG) and are suitable for the short term development of a demonstrator. Other algorithms use with later versions using advanced coding techniques and will contribute for the definition of future standards. Rate Compatible Punctured Convolutional (RCPC) codes and combined source-channel coding have been introduced into the video coding task.

Further achievements are expected in the areas of:

- Non Compatibility of Video Coding under UMTS
- Development of Video Front-end Hardware
- Development of Video Codec Hardware
- Control Driver Functions for a Video Codec
- Implementation of the Video and Audio Algorithms
- Platform Development

Expected Impact

Project results will contribute to standardization activities within ISO, CCITT and ETSI. Improvements made to existing coding algorithms could influence standardisation of the compatible coder (recommendations H.261, MPEG, JPEG).

Results will be useful to designers and manufacturers of services and terminals, by providing basic design guidelines and human recommendations for mobile environments. In this way, they may accelerate the penetration of mobile handheld terminals in the market, as an attractive form of personal communication.

Participants

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APPENDIX 5.7

R2072:
MOBILE AUDIO-VISUAL TERMINAL (MAVT):
CHANNEL CODING ASPECTS

ITU-TS SG15 Rapporteur
for Very Low Bitrate Visual Telphony

Document LBC-93-07?

R2072 Mobile Audio Visual Terminal: Channel Coding Aspects

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Abstract

The main technical objectives of MAVT are the development of robust video and audio coding algorithms for transmission of moving (still) video and associated audio in DECT and UMTS. Due to a variety of channel impairments encountered in case of a wireless mobile propagation medium additional countermeasures have to be undertaken to guarantee the target transmission quality. This paper includes corresponding error control strategies under investigation in the context of the RACE MAVT project.

I. Introduction

A new challenging field of research is the area of very low bit rate video source coding with data rates of px8 kb/s (n=1..8) in connection with a mobile wireless transmission such as in DECT and the future UMTS. Corresponding H.261 non-compatible and compatible source coding algorithms have been developed in the context of the MAVT project. In general low bit rate source coding algorithms are very sensitive to channel errors due to the high compression factors. Furthermore, a target network may only guarantee a specific average transmission quality (network dependent), which will not suffice in the consider application (e.g. DECT). Therefore, an additional service dependent error control strategy must be applied, which should consider the underlying channel characteristics as well as the sensitivity of the source against channel errors or error events.

Section II includes some aspects related to the network and transmission channel characteristics in case of the target networks, with special emphasis on DECT. An efficient channel coding design must consider the characteristics of the underlying discrete channel, which determines the error process. Results obtained through a descriptive modelling approach in case of a data transmission in DECT with 32 kb/s are presented.

In the context of MAVT two source coding approaches are considered, namely a H.261 non-compatible hybrid algorithm and H.261 related algorithms with different information data rates. Corresponding forward error correction strategies are presented in Section III, where a-priori knowledge about the bit sensitivity of the source codec is incorporated in the design of a non-uniform error protection scheme.

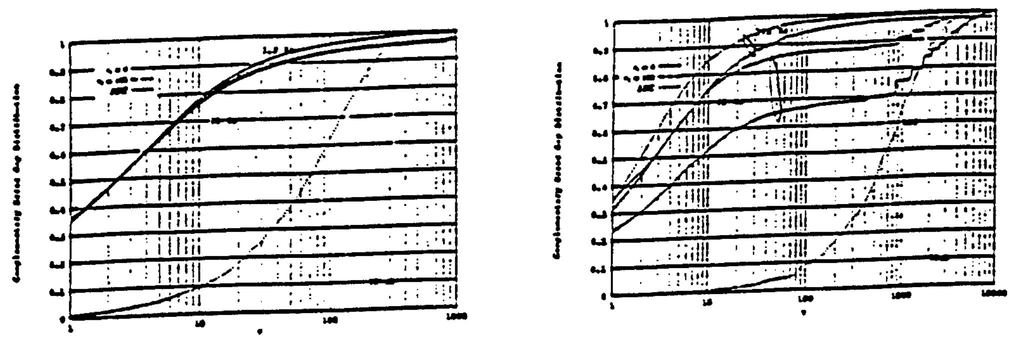
II. Channel Characteristics

A general time-varying multipath channel introduces time and frequency dispersion in the transmitted, whereby two interference effects can be described. In case of a pure time selective or frequency dispersive propagation channel, the field strength of the received envelope exhibits extreme variations, i.e., signal fading, which are inversely proportional to the maximum Doppler frequency $f_{D_{max}}$. For low signal levels, i.e., a low signal to noise ratio, this implies error bursts in case of a data transmission due to the channel memory. For high SNRs the effects due to the random phase predominates yielding an irreducible error floor.

A pure frequency selective or time dispersive propagation channel is characterized by a frequency dependent transfer function, where the variations are inversely proportional to the underlying delay spread τ_v. Due to the nonideal spectral characteristic, the transmitted signal is linearly distorted, which implies intersymbol interference (ISI) in case of a data transmission. The general case is characterized by a superposition of the above two effects.

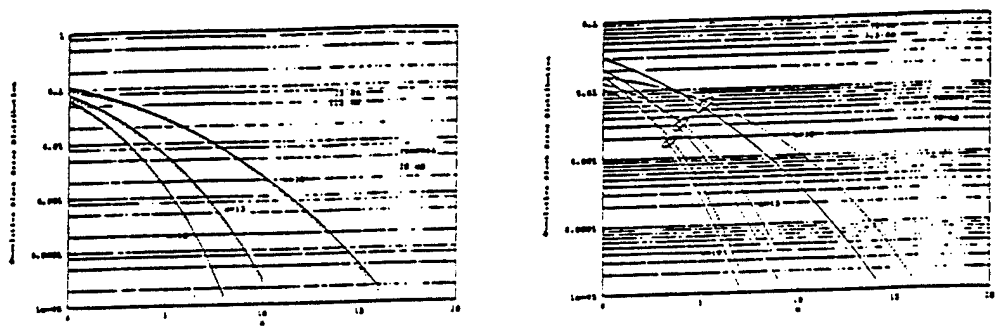
The statistical structure of the resulting error process is completely determined by the underlying discrete channel. Relevant statistical parameters can be determined from corresponding error sequences either by direct processing (descriptive modelling) or from a parameterized mathematical model (generative modelling). The error process can be equivalently described in terms of successive gaps of length g_i, where a gap is defined as a string of v-1 consecutive error free symbols between two error symbols and having a length equal to v. The gap process represents a realization of a stochastic process and therefore it can be described by corresponding statistics such as the gap density and cumulative distribution, whereby in case of a so called renewal process, i.e., uncorrelated successive gap lengths, the gap process is uniquely defined by the single error gap density $f(v)=Pr(g_i=v)$ or distribution $F(v)=Pr(g_i>v)$, $\forall\ v \in \mathbb{N}$.

A relevant parameter for code design in case of random error correction is the block error probability $P(m,n)$=Pr(number or errors=m, block of length n). Herewith one obtains the resulting error rate in case of a block code capable of correcting t errors directly. In case of burst error correction, the relevant parameter is the burst error probability $Q(l,n)$=Pr(burst length=l, block of length n), where a burst is the length starting from the first error to the last error irrespective of the structure in between. For bit interleaving the relevant parameter is the autocovariance coefficient $\rho(\kappa)=cov(e_0,e_\kappa)/var(e)$ of the error process. The interleaving degree should be chosen equal to the value of κ for which $\rho(\kappa)$ is zero or a minimum. Further relevant statistics, as well as the ones discussed above, can be derived directly from the corresponding gap process and are presented in [1]. The following figures represent error profiles evaluated from corresponding error sequences in case of a DECT transmission link and a general time-variant multipath channel with a data rate of 32 kb/s. The propagation channel is characterized by the classical Doppler spectrum and a one sided exponential power profile.



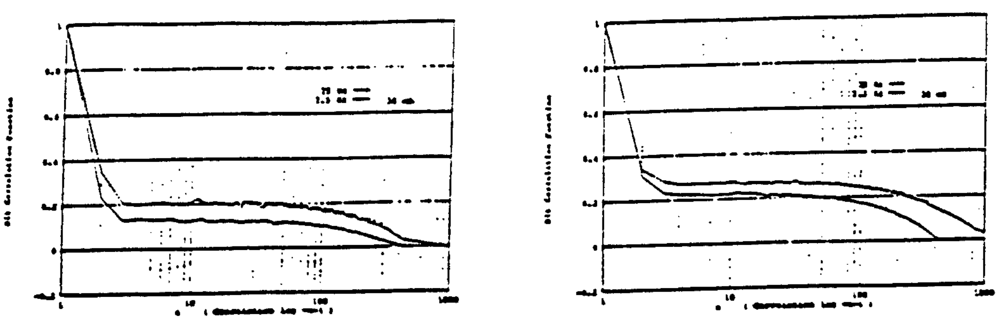
a b

Fig.1 Anti-complementary error gap distribution 1-F(ν)
(f_{Dmax}=2.5,25 kHz; τ_s=0, 100 ns) a SNR=20 dB. b SNR=30 dB



a b

Fig. 2 Cumulative block error distribution P(>m,n)
(f_{Dmax}=2.5,25 Hz; τ_s=0 ns) a SNR=20 dB. b SNR=30 dB



a b

Fig. 3 Autocovariance coefficient ρ(κ)
(f_{Dmax}=2.5,25 Hz; SNR=30 dB) a τ_s=0 ns. b τ_s=100 ns.

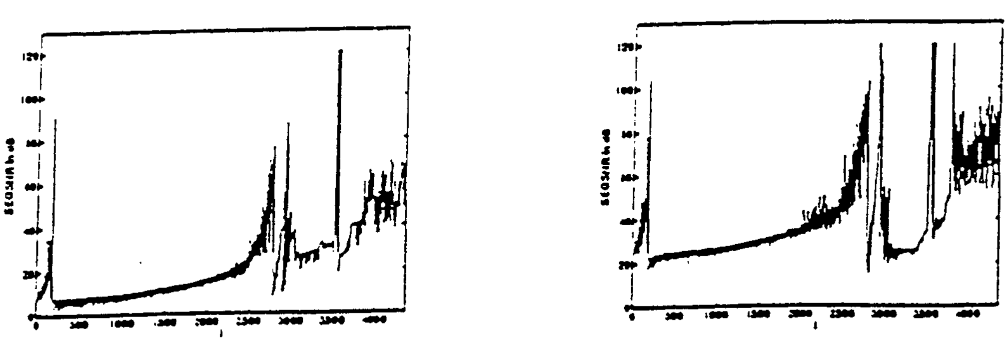
III. Combined Source-Channel Coding

A. Source coding

There are two video source coding approaches which are followed in the context of MAVT, namely a H.261 noncompatible hybrid codec and several H.261/MPEG related algorithms [2]. The considered source data rates in case of DECT are 8, 16 and 24 kb/s for the noncompatible and 16 and 24 kb/s for the compatible code. The frame rate equals 6.25 Hz, while the spatial resolution is QCIF in both cases. The associated CELP speech codec is characterized by a total data rate of 8 kb/s including channel coding. The compound transmission in DECT includes additional control and synchronization information and will be accomplished by using one slot of the TDMA frame, i.e., with a gross data rate of 32 kb/s. Both coding approaches will be applied in case of narrowband UMTS, while in the broadband case the H.261 compatible scheme alone will be considered. In this application higher data rates (>32 kb/s) are allowed.

B. Error control strategies

Due to compatibility restriction a uniform error protection (block codes) for the information part adapted to the corresponding application will be followed in case of the H.261 compatible scheme. Key aspects are the selection of an appropriate frame alignment signal (FAS) inclusive error protection and the necessary error protection of the bit rate allocation signal (BAS) in case of the target wireless mobile transmission. In case of the noncompatible video source codec an unequal random error protection scheme will be adopted. The design of the proper individual channel codes considers a-priori knowledge about the sensitivity of the source information against channel errors. Reference to figure 4 reveals, that the individual bits i at the output of the source encoder exhibit a significant difference with respect to their segmental signal to noise ratio (SEGSNR) in case of single independent channel errors.



a　　　　b

Fig. 4 Bit sensitivity of the H.261 noncompatible hybrid video codec with 16 kb/s. a Luminance. b Chrominance

This fact suggests the application of different error protection levels, which can be realized efficiently through rate compatible punctured convolutional (RCPC) codes [3]. A further approach which is under investigation is a concatenated scheme, composed of an outer block codec and an inner RCPC codec. In both cases a convolutional interleaver will be applied for randomizing the underlying error events. In general the optimization of the proper error correction scheme represents a compromise between affordable redundancy, resulting delay and computational burden. On the receiver side several strategies for improving the system performance can be applied. Current investigation are concerned with soft-decision video decoding and efficient error concealment methods.

References

[1] P. Crespo, R. Mann Pelz, J. Cosmas: Channel Error Profile for DECT, R2072/TEL/1.2/DS/R/001/b1, July, 1992

[2] D. Lappe et al.: Video Coding Algorithm px8 kb/s for UMTS and DECT, R2072/BOS/2.1/DS/R/009/b1, January, 1992

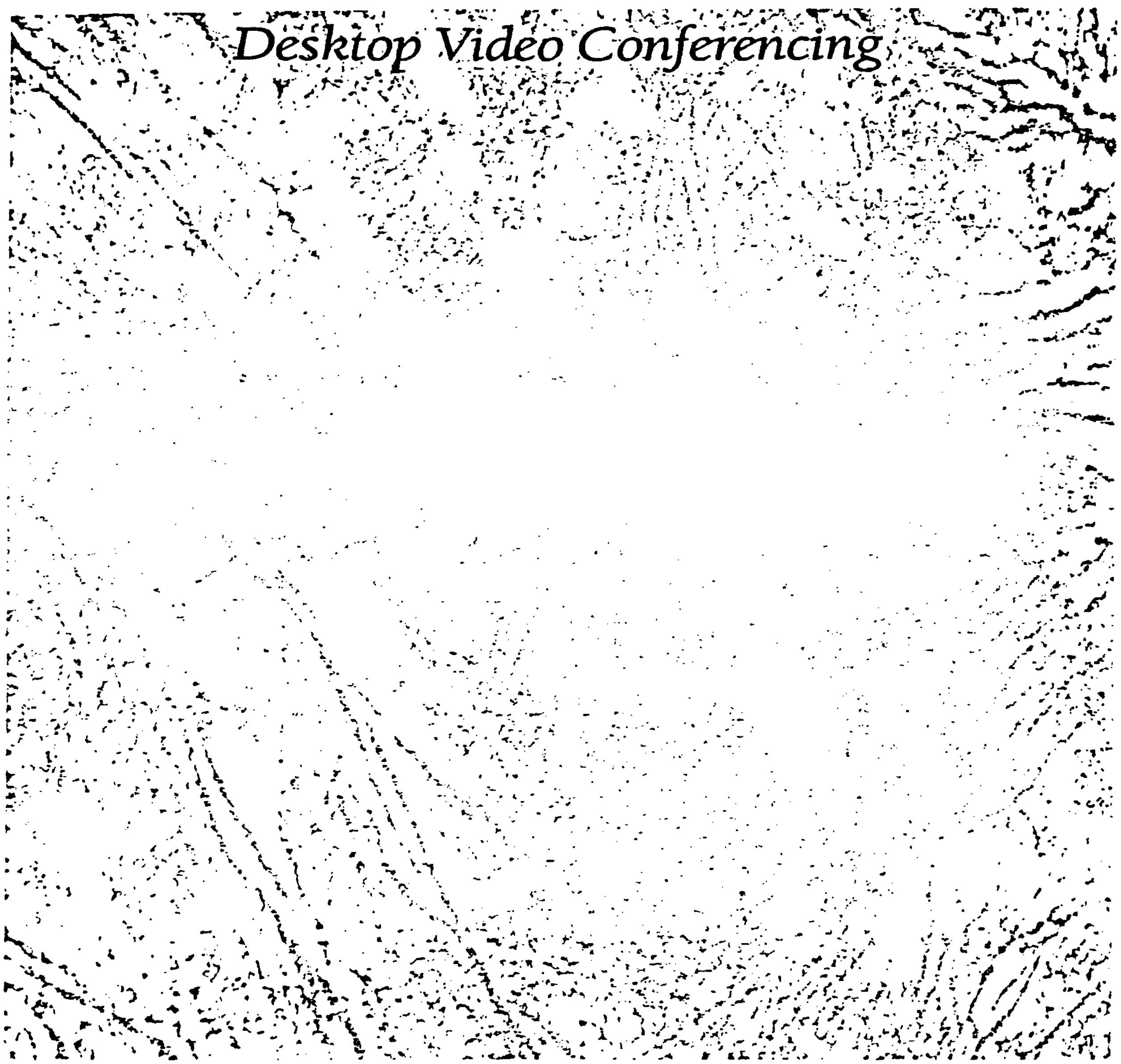
[3] J. Hagenauer: Rate-Compatible Punctured Convolutional Codes (RCPC Codes) and their applications, IEEE Trans. on Comm., COM-36, pp. 389-400, 1988

APPENDIX 6.1

PICTURE WINDOW

PictureWindow™

Desktop Video Conferencing



BBN Systems and Technologies
10 Moulton Street
Cambridge, MA 02138

2. What is PictureWindow?

BBN's PictureWindow system brings multiparty, wide-area video conferences directly onto your workstation screen with minimal additional hardware. PictureWindow gives you personal video conferencing capability through the use of BBN software, Sun's inexpensive VideoPix frame-capture board, and a video camera on your existing color SPARCstation.

PictureWindow compresses and decompresses video entirely in software, transmits it using standard IP protocols, and displays it in multiple windows under OpenWindows™ or the X Window System. It operates with equal ease between offices and between continents over Internet Protocol (IP) based wide-area networks, and it can be used in either point-to-point or multicast modes.

What PictureWindow Does

In PictureWindow, video from each conferee appears in a separate workstation window. Participants may join or leave the conference at any time; new windows appear as participants join and disappear when they leave. A user can only participate in one video conference at a time, but the number of participants in the conference is limited only by workstation and network speed.

Initiating or joining a conference is simple: you run the PictureWindow program and press the *Connect* ... button. PictureWindow will display your personal video conference address book. Select a person to confer with and confirm the connection; if the person you have selected accepts your call, video images from their workstation will appear within a few seconds and your image will automatically be displayed at their location. To add more conferees, just press *Connect*... again. You can terminate a single connection by pressing the *Hang up* button in its video window. You can terminate the entire conference by clicking on the *Hang up all* button in the main PictureWindow window.

PictureWindow video windows measure 320x240 pixels with 16 levels of gray. Refresh rate depends upon system and network load, but is typically between 3 and 6 frames per second. A SPARCstation™ 1, 1+, 2, IPC, or IPX will support video conferences with six to eight participants. PictureWindow easily coexists with other applications on the workstation, so users can confer over documents using other tools while participating in a video conference.

PictureWindow sends compressed video in UDP/IP datagrams and functions in both local and wide-area network environments. The actual network bandwidth used by any one conferee depends on the amount of motion in the supplied video, and the image quality desired by the viewers. In two-way conferences, each conferee can selectively adjust the quality and compression parameters for the image they are viewing. PictureWindow functions best when network paths with at least 256 kilobits/second are available. Nevertheless, network paths as slow as 56 kilobits/second can be used by decreasing the frame rate and increasing the acceptable image error.

PictureWindow System Requirements

PictureWindow currently runs on Sun Microsystems® SPARCstations or fully compatible workstations equipped with 8- or 24-bit color frame buffers. It requires a local X server with the MIT shared-memory extension, such as MIT X11R5, OpenWindows 2.0, or OpenWindows 3.0. Transmit capability requires the installation of the Sun VideoPix card; however, receive-only copies of the software are available that do not require any additional hardware to be added to your Sun SPARCstation.

Video input can come from any color or black-and-white television source, such as a camera, camcorder, or VCR. Available video input formats are NTSC and PAL video, and they can be supplied using composite or S-Video input connectors. Sources with lower background noise will yield better video compression ratios.

The required hardware to run PictureWindow is as follows:

- Sun SPARCstation 1, 1+, 2, 10, IPC, or IPX workstation
- 8-bit or 24-bit color or grayscale frame buffer
- 16 Meg main memory (minimum)
- Sun VideoPix card
- Black-and-white or color, NTSC or S-Video camera
- Ethernet with TCP/IP protocol stack running (standard Sun configuration)
- Access to CD-ROM drive to load VideoPix software driver
- Access to floppy disk drive to load PictureWindow software (standard on all above workstations).

The operating system software requirements are:

- SunOS™ 4.1.1 or later[1]
- Sun VideoPix device driver
  (i.e., system reports:
  `vfc0 at SBus slot 1 0x0` on booting)
- IPCSHMEM option enabled *(Note: the GENERIC kernel configuration that came with your Sun has this option enabled, but the GENERIC_SMALL kernel configuration does not).*

You also need to run an X Window System-compatible windowing environment. The requirements for this software are:

- X Window System compatibility
- An X Window Server that exports an 8-bit *PseudoColor* visual
- The MIT-SHM extension that permits memory-mapped X windows.

OpenWindows 2.0, OpenWindows 3.0, X11 Release 4, and X11 Release 5 will all satisfy the requirements for the windowing software when run on 8-bit or 24-bit color displays.

[1] At present, Picture Window is not supported under Solaris® versions 2.0 or later.